KerioOperator

ADMINISTRATOR GUIDE

Find out how to configure Kerio Operator in different environments and how to set up advanced features.

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Contents

1 Introduction to Kerio Operator

Kerio Operator is a VoIP based phone system that provides enterprise-class voice and video communication capabilities for small and mid-sized businesses globally. Easy to administer and flexible to deploy, as a software appliance, a virtual machine, a hardware appliance, or a cloud solution.

Kerio Operator brings support for high-quality codecs, such as Opus for voice and H.264 for video.

The automatic provisioning feature brings fast setup for various phone systems, such as Cisco, Grandstream, Polycom and Snom phones.

Its advanced security technologies keep telephone hackers out, prevent misuse and ensure the privacy of your users and those they call. Kerio Operator continually monitors for anomalous behavior, detects and prevents break-in attempts and supports call encryption.

Kerio Operator also includes various call handling features which are used on a daily basis, such as auto attendant scripts, advanced call forwarding, call pickup, Busy Lamp Field (BLF), and many more.

You can stay in control of all your Kerio Operator appliances through Kerio's centralized web interface - MyKerio.

This help system includes technical information about how to deploy, use, configure and troubleshoot Kerio Operator.

Futher reading:

» Getting started with Kerio Operator

» Logging into Kerio Operator Administration

2 Getting started

Want to try out Kerio Operator? This topic provides a quick list of actions to help you set it up.

1	**Prepare external connectivity** Kerio Operator requires connectivity with a telecommunications service provider (TSP) or an Internet telephony service provider (ITSP) to make and receive external calls. Refer to the supported phone cards and tested SIP providers on the Kerio website.
2	**Install Kerio Operator** You can install Kerio Operator as a hardware, software, or virtual appliance. All installation types use a built-in operating system that you manage through the web administration. Refer to the technical specifications for the requirements of each option. For more information, refer to Installing Kerio Operator (page 9).
3	**Access the Kerio Operator interface** You can administer Kerio Operator directly on the network using a web browser by opening a secure connection to the IP address or hostname of your server. For more information, refer to Logging into Kerio Operator Administration (page 19).
4	**Activate Kerio Operator** When launching the web administration interface for the first time, run through the configuration wizard to activate essential settings. For more information, refer to Configuration wizard (page 19).
5	**Configure Kerio Operator on the network** To communicate on the network, assign network parameters to Kerio Operator from Configuration > Network. Configurable items include NAT, domain name server address for resolving names, IP address, gateway, and subnet for routing to the Internet and local networks.
6	**Add extensions and user accounts** To manage calls, you need to create extensions and assign them to your users and phones. Create user accounts and assign extensions to them.
	Connect to a telephone service provider Telephone service using PRI/BRI, POTS, or Euro-ISDN require physical infrastructure and a hardware interface with specific configuration for each type of service. For more information, refer to Configuring standard phone interfaces (page 270). Telephone service using a SIP provider requires Internet access. Kerio Operator uses a virtual SIP interface to connect to the SIP provider. For more information, refer to Connecting to VoIP service providers (page 25).
	Deploy user phones You can manually provision phones through a software interface, or Kerio Operator can automatically provision phones. Click here for a list of phones supporting automatic provisioning with Kerio Operator. For more information, refer to Configuring automatic phone provisioning (page 170). You can also use Kerio Operator Softphone app with mobile device or Kerio Phone for desktop operating systems. You can also use third-party softphones with Kerio Operator. For more information go to http://go.gfi.com/?pageid=operator_help#cshid=924

2.1 System requirements for Kerio Operator

You can find detailed and always up-to-date **system requirements** for Kerio Operator on our website:

Kerio Operator System Requirements

2.2 Installing Kerio Operator

2.2.1 Product Editions

Edition	Description
Software Appliance	Kerio Operator Software Appliance is an all-in-one package of Kerio Operator which also includes a special operating system. Designed to be installed on a computer without an operating system, this edition is distributed as an installation disc. Software Appliance cannot be installed on a computer with another operating system and it does not allow to install other applications. For more information, refer to Kerio Operator Software Appliance (page 9).
VMware Virtual Appliance	A virtual appliance designed for use in VMware products. VMware Virtual Appliance is a Software Appliance edition pre-installed on a virtual host for VMware. The virtual appliance is distributed as OVF and VMX. For more information, refer to Kerio Operator VMware Appliance (page 9).
Kerio Operator Box	Hardware device ready for network connection. There are two types which differ in performance. For more information, refer to Kerio Operator Box (page 10).

2.2.2 Kerio Operator Software Appliance

For Kerio Operator system requirements, refer to the Kerio Operator product pages.

You obtain Kerio Operator as a standard ISO image which you need to burn on a CD. Boot from this CD and install the Kerio Operator operating system. The Kerio Operator application is also installed during the process.

How to connect Kerio Operator Software Appliance to network

After booting the system, a console with the IP address for Kerio Operator is displayed.

If you use a DHCP service on your network, Kerio Operator will be assigned an IP address automatically and will connect to the network. If you do not use or do not wish to use DHCP for Kerio Operator, you have to set the IP address manually.

The current network configuration is displayed (and can be changed) in the Kerio Operator console in section `Network Configuration`. To set a static network address:

1. Select the `Assign static IP address` option in the console menu.

2. In the network interface on which the PBX should communicate, select the `Assign static IP address` option and enter the IP address, subnet mask and IP addresses of gateway and DNS server.

If you know the DNS name of the PBX, you can connect to it and configure it via web interface.

> **IMPORTANT**
> Immediately after you connect Kerio Operator to the network, we recommend to read topic concerning the security measures. Meeting security principles for Kerio Operator operation is extremely important. If the PBX is not protected by a firewall and supporting security rules, your internal telephone extension can be misused which may result in unexpected financial costs.

2.2.3 Kerio Operator VMware Appliance

For supported VMware product versions, check http://www.kerio.com/operator/requirements/

Use an installation package in accordance with the type of your VMware product:

» For products VMware Server, Workstation, Player and Fusion, download the compressed VMX distribution file (*.zip), unpack it and open the file with extension .vmx.

» You can import a virtual appliance directly to VMware ESX/ESXi from the URL of the OVF file — for example: http://download.kerio.com/dwn/operator/ kerio-operator-appliance-2.3.0-2500-vmware.ovf. VMware ESX/ESXi automatically downloads the OVF configuration file and a corresponding disk image (.vmdk).

If you import virtual appliance in the OVF format, bear the following specifics in mind:

» In the imported virtual appliance, time synchronization between the host and the virtual appliance is disabled. However, Kerio Operator features a proprietary mechanism for synchronization of time with public Internet time servers. Therefore, it is not necessary to enable synchronization with the host.

» Tasks for shutdown or restart of the virtual machine will be set to default values after the import. These values can be set to hard shutdown or hard reset. However, this may cause a loss of data on the virtual appliance. Kerio Operator VMware Virtual Appliance supports so called Soft Power Operations which allow to shut down or restart hosted operating system properly. Therefore, it is recommended to set shutdown or restart of the hosted operating system as the value.

For more information, refer to How to connect Kerio Operator Software Appliance to network (page 9).

2.2.4 Kerio Operator Box

For currently supported Kerio Operator Box configurations, refer to the Kerio Operator product pages.

For detailed information on connecting the device into the network, see the Kerio Operator Box 1000/3000 Series and Kerio Operator Box V300 installation guides.

How to connect to the hardware box from the network

Upon the first start, the appliance has a static IP address set to 10.10.10.1 on ethernet port 1. There are two ways to change the configuration:

» In the console — use an Ethernet cable to connect to the console. In the console menu, select the **Network Configuration** option and change the configuration.

» In the administration interface in section **System**. To connect to Kerio Operator, set the following TCP/IP parameters on your computer:

a. IP address: 10.10.10.2

b. Subnet mask: 255.255.255.0

To shut down the appliance:

1. Connect to Kerio Operator via the console and select the Shutdown command.

2. Kerio Operator series 1000 will shut down. Kerio Operator series 3000 will stop the server, however, the physical appliance stays switched on. Wait until you are not able to connect to Kerio Operator via Kerio Operator administration and turn the appliance off using the **pwr** button on the appliance.

2.3 Hardware appliances

This section describes deployment and configuration for hardware appliances.

2.3.1 Kerio Operator Box V300

Learn how to safely install and implement Kerio Operator Box V series PBX appliances.

General Safety Instructions

During installation follow these security instructions:

» The appliance should be placed on a flat surface.

» Do not attempt to open or disassemble the appliance for any reason.

» Strictly follow the installation instructions.

» Do not place the appliance near a heat source.

» Place the appliance in a ventilated space, making sure that the appliance fans and vents are unobstructed at all times.

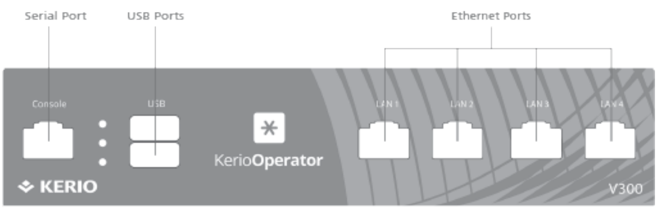
» Do not expose the appliance to liquids of any kind. In the event of liquid intrusion, unplug the appliance immediately.

» Verify that the voltage and frequency of the power socket matches the values printed on the power adapter before plugging in the appliance. Use only the power adapter supplied with the appliance.

» Do not place any items on top of the power cable; keep the power cable away from walkways or other areas where it could pose a tripping hazard.

Appliance Description

Kerio Operator Box V300 is a Sub-1U table mountable appliance.



Feature	Description
Serial port	Used for connecting to a console with a serial cable
USB ports	Input for USB devices
Ethernet network ports	Used for connecting to the Internet and the LAN with an Ethernet cable

Kerio Operator Box Installation and Configuration

Once a suitable place has been located for the appliance and it has been plugged into a power outlet according to the safety instructions, it is time to connect it to the network and configure settings.

1. Connect Ethernet port number 1 to the network using an Ethernet cable.

> **NOTE**
> Alternatively, you can use port number 2 which includes a DHCP client.

2. Turn on the appliance.

3. On the computer you want to use for the Kerio Operator configuration, set **IP address** to `10.10.10.2` and **Subnet mask** to `255.255.255.0`. Setting the default gateway and DNS servers is not necessary for the Kerio Operator configuration.

4. The Kerio Operator PBX is configured through the Kerio Web Administration interface. Open a web browser and connect to the Kerio Control Administration web interface using the **https://10.10.10.1/admin** URL.

5. Ignore the SSL certificate warning.

6. Follow the instructions provided by the wizard and configure the appliance.

> **NOTE**
> For troubleshooting purposes, you can use the serial port to connect the console to the device. For more information, refer to Connecting to Kerio hardware appliances with a serial console (page 14).

Additional Information

For further assistance with configuration please refer to additional documentation at:

- https://manuals.gfi.com/en/kerio/operator/content/home.htm

For online and community based support resources please visit:

- http://www.kerio.com/support

2.3.2 Kerio Operator Box 1000/3000 Series

Learn how to safely install and implement Kerio Operator Box 1000 and 3000 Series PBX appliances.

General Safety Instructions

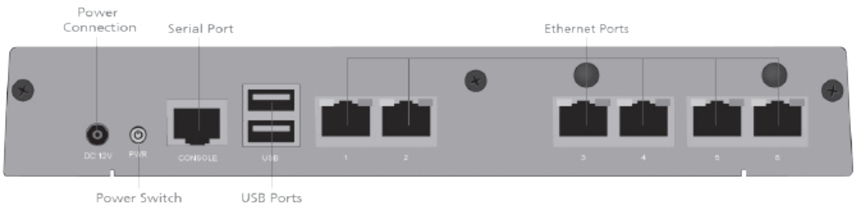
During installation follow these security instructions:

- The appliance should be placed on a flat surface or securely mounted horizontally in rack enclosure.
- Do not attempt to open or disassemble the appliance for any reason.
- Strictly follow the installation instructions (see section 4).
- Do not place the appliance near a heat source.
- Place the appliance in a ventilated space, making sure that the appliance fans and vents are unobstructed at all times.
- Do not expose the appliance to liquids of any kind. In the event of liquid intrusion, unplug the appliance immediately.
- Verify that the voltage and frequency of the power socket matches the values printed on the power adapter before plugging in the appliance. Use only the power adapter supplied with the appliance.
- Do not place any items on top of the power cable; keep the power cable away from walkways or other areas where it could pose a tripping hazard.

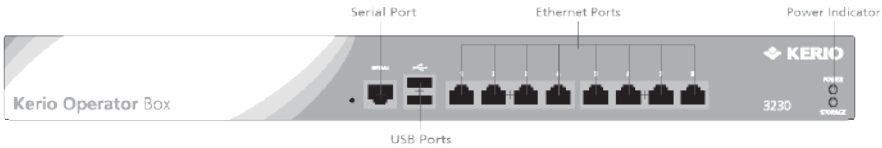
Device Description

Kerio Operator Box types:

- Kerio Operator 1000 Series — Sub 1U table mountable appliance (see figure 1).
- Kerio Operator 3000 Series — 1U rack mountable appliance (see figure 2).



Screenshot 1: Figure 1 Kerio Operator Box 1220



Screenshot 2: Figure 2 Kerio Operator Box 3230

Feature	Description
Serial port	Used for connecting to a console with a serial cable
USB ports	Input for USB devices
Ethernet network ports	Used for connecting to the Internet and the LAN with an Ethernet cable

Kerio Operator Box Installation and Configuration

Once a suitable place has been located for the appliance and it has been plugged into a power outlet according to the safety instructions, it is time to connect it to the network and configure settings.

1. Connect Ethernet port number 1 to the network using an Ethernet cable.

> **NOTE**
> Alternatively, you can use port number 2 which includes a DHCP client.

2. Power the device with the power switch. For 3000 series, the power switch is located in the rear of the device.

3. On the computer you want to use for the Kerio Operator configuration, set **IP address** to `10.10.10.2` and **Subnet mask** to `255.255.255.0`. Setting the default gateway and DNS servers is not necessary for the Kerio Operator configuration.

4. The Kerio Operator PBX is configured through the Kerio Web Administration interface. Open a web browser and connect to the Kerio Control Administration web interface using the **https://10.10.10.1/admin** URL.

5. Ignore the SSL certificate warning.

6. Follow the instructions provided by the wizard and configure the appliance.

> **NOTE**
> Alternatively, you can use the serial port to connect the console to the device. After the server starts, you can get information about the actual network configuration or you can use the console to restart or turn off the appliance. Set your terminal application in the following mode: 9600, 8, N, 1.

Additional Information

For further assistance with configuration please refer to additional documentation at:

- https://manuals.gfi.com/en/kerio/operator/content/home.htm

For online and community based support resources please visit:

- http://www.kerio.com/support

2.3.3 Setting Up Kerio Operator Box 1220 and 3230

There are two Kerio Operator Box models available:

- **Kerio Operator Box 1220** - A small desktop appliance featuring six Gigabit Ethernet ports.
- **Kerio Operator Box 3230** - A 1U rack-mount appliance featuring eight Gigabit Ethernet ports.

> **WARNING**
> Kerio Operator Box 3230 is intended primarily for server rooms due to noisy performance.

For more information, refer to Kerio Operator Box 1000/3000 Series (page 12).

2.3.4 Connecting to Kerio hardware appliances with a serial console

Connecting to the Kerio Control hardware appliance through a serial console can help you in the following cases:

- Broken network access to the hardware appliance due to configuration mistakes or network hardware issues (both from the box and network switch sides)
- Direct access to the Linux shell
- You need to see the boot sequence from the hardware appliance
- Access to BIOS

Setting a communication through a serial console

The connection uses these settings:

- Speed: 9600
- Data bits: 8
- Stop bit: 1
- Parity: none
- Flow control: none

Accessing BIOS

The connection uses these settings:

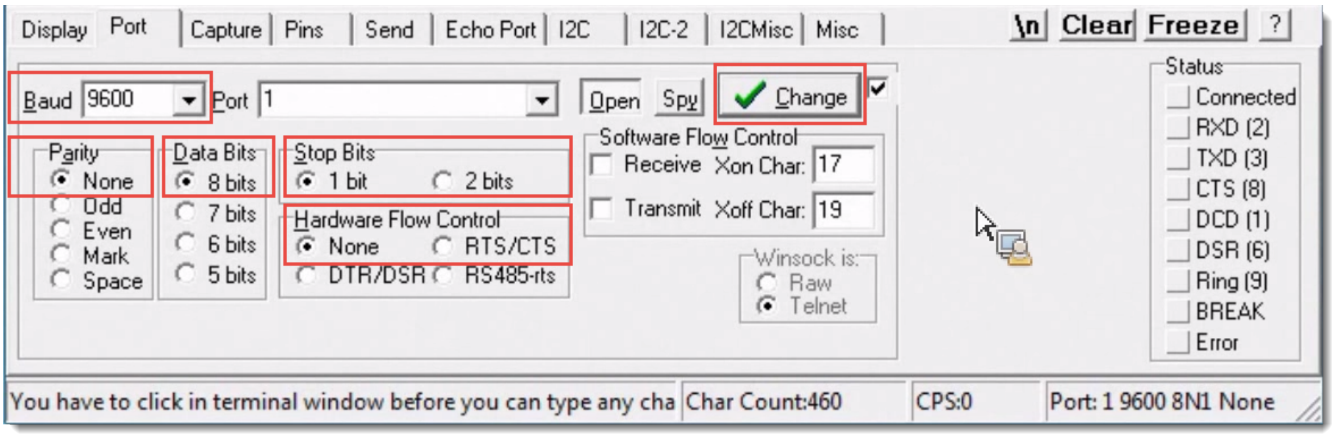
» Speed: 115200
» Data bits: 8
» Stop bit: 1
» Parity: none
» Flow control: none

Use the instructions for your operating system to create these settings:

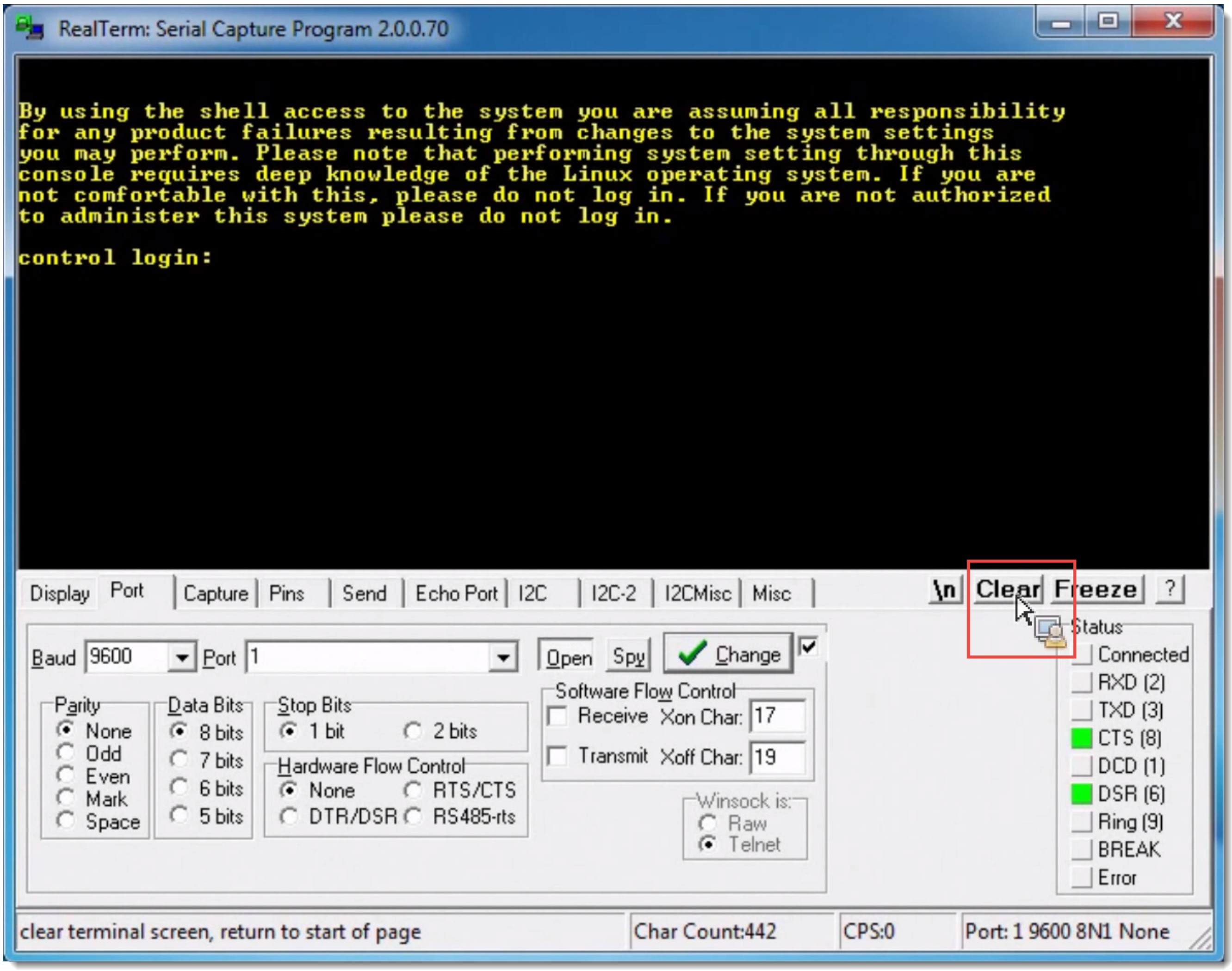
Windows

To connect to the hardware appliance, you need a special application such as PuTTY or RealTerm. Here are the steps for RealTerm:

1. Install RealTerm on your computer.
2. Attach the serial cable to the hardware appliance and to your PC.
3. Run RealTerm.
4. On the **Display** tab, select **ANSI**.
5. Click the **Port** tab and make the following selections there:
    - Baud: 9600
    - Parity: None
    - Data Bits: 8
    - Stop Bits: 1
    - Hardware Flow Control: None
6. Click **Change**.



Before logging on to your hardware device, click **Clear**.



Now, you can log in to your hardware device as root. Use the admin password for verification.

Linux

To connect to the hardware appliance, you need a special terminal software such as minicom. Here are the steps for minicom:

1. Install the minicom application.

2. Type the following command at the shell prompt: `$minicom -s`

3. In the menu, select **Serial port setup**.

```
+-----[configuration]------+
| Filenames and paths      |
| File transfer protocols  |
| Serial port setup        |
| Modem and dialing        |
| Screen and keyboard      |
| Save setup as dfl        |
| Save setup as..          |
| Exit                     |
| Exit from Minicom        |
+--------------------------+
```

4. Type **A**.

5. In the **A** section, type the interface: **TTYS0**. If you use an USB-to-serial adapter, select **USB** instead.

6. Press **Enter**.

7. Type **E**.

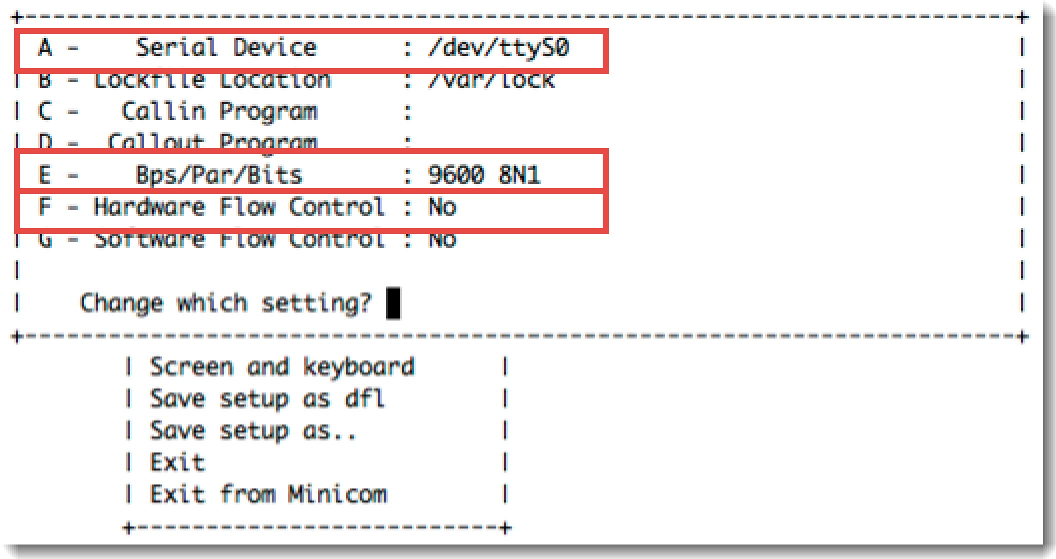
8. In the **E** section, type **CQ**.C: 9600 baud, Q: 8 bits, parity: none, stop bit: 1.

```
+-----------------+---------[Comm Parameters]----------+----------------+
| A -    Serial De|                                    |                |
| B - Lockfile Loc|    Current:  9600 8N1              |                |
| C -   Callin Pro| Speed          Parity      Data    |                |
| D -  Callout Pro| A: <next>      L: None     S: 5    |                |
| E -    Bps/Par/B| B: <prev>      M: Even     T: 6    |                |
| F - Hardware Flo| C:   9600      N: Odd      U: 7    |                |
| G - Software Flo| D:  38400      O: Mark     V: 8    |                |
|                 | E: 115200      P: Space            |                |
|   Change which  |                                    |                |
+-----------------| Stopbits                           |----------------+
        | Screen a| W: 1           Q: 8-N-1            |
        | Save set| X: 2           R: 7-E-1            |
        | Save set|                                    |
        | Exit    |                                    |
        | Exit fro| Choice, or <Enter> to exit?        |
        +---------+------------------------------------+
```

9. Press **Enter**.

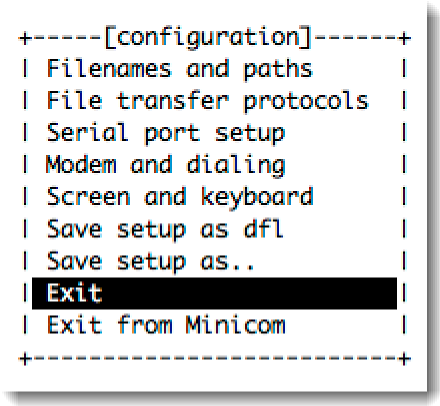
10. Type **F** and set it to **No**.

11. Press **Enter** to save the configuration.



12. Return to the main menu.

13. Select **Exit**.



Now, you can log in to your hardware device as root. Use the admin password for verification.

OS X

To connect to the hardware appliance, you need:

- USB to Serial adapter with the FTDI chipset directly supported by OS X.
- Special terminal software such as CoolTerm.

Here are the steps for CoolTerm:

1. Put the serial cable to the hardware appliance and also to your Mac with the USB to Serial adapter.

2. Open CoolTerm.

3. In the **Serial Port** section, select the USB adapter as port.

4. Baudrate: 9600.

5. Data Bits: 8.

6. Parity: none.

7. Stop Bits: 1.

8. Flow Control: no selection.

9. Click **Connect**.

Now you can log in to your hardware device as root. Use the admin password for verification.

2.4 Logging into Kerio Operator Administration

We recommend to use the supported browsers to connect to Kerio Operator Administration. For the list of the browsers, refer to the Kerio Operator product pages.

Kerio Operator Administration is currently localized into several languages. Select yours in the top right corner of the interface. The default language is set according to your browser language settings.

2.4.1 How to login

Before you login for the first time, make sure you have:

- DNS name of the server with Kerio Operator.
- Supported browser

To login, enter the DNS name of the computer with Kerio Operator: `kerio.operator.name/admin`

Administration runs solely via the HTTPS protocol on port 4021. The address is automatically redirected to: `https://kerio.operator.name:4021/admin`

> **NOTE**
> If the PBX is located behind firewall, HTTPS on port 4021 must be enabled.

If the URL is entered correctly, your browser displays a warning about a SSL certificate. After the installation, Kerio Operator creates a certificate which is not signed by a trusted certificate authority — it is a self-signed certificate (for more information, read topic about the SSL certificates). Since you know the certificate can be trusted, you can add the security exception and continue to a login page.

Configuration wizard

When you connect to the PBX for the first time, a configuration wizard to do the necessary configuration. Here are those settings:

1. Set the configuration wizard language.

2. Accept the Kerio Operator license agreement.

3. Set a password for the administration account (be sure to remember the password, you will need it to login to the PBX).

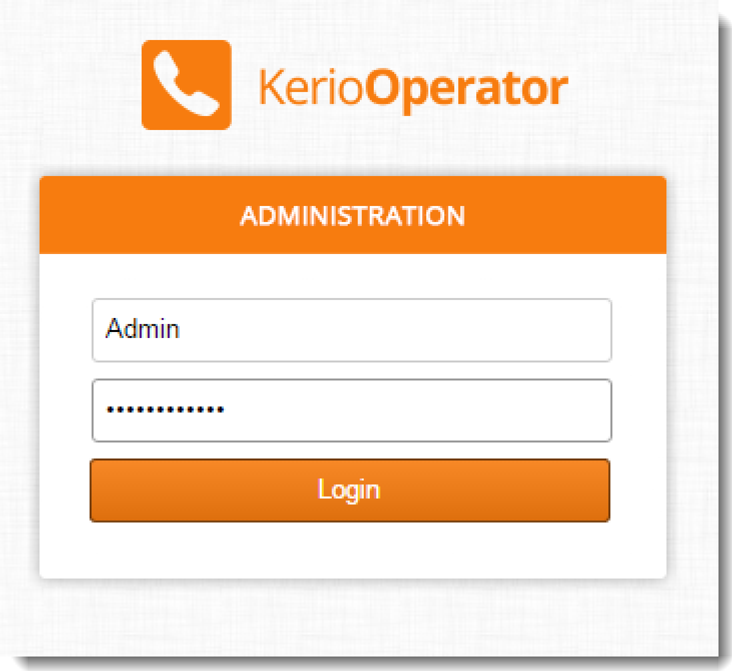
> **NOTE**
> This admin password is synchronized with password of user `root` in the operating system where Kerio Operator is installed (Kerio OS).

4. Set the time zone of Kerio Operator (requires a restart of the PBX).

5. Set the PBX language for communication with you and other users (warnings, auto attendant scripts, voicemail, etc.).

6. Configure the first extension number. If you use phone provisioning, extensions will be created automatically beginning with the number you enter here.

After successful configuration, the login page is displayed. Enter the username and password you created earlier.



To change the password, use the following steps:

1. Login to Kerio Operator using the HTTPS protocol (e.g. `https://operator.company.com/admin`)

2. Open the **Configuration > Users** section.

3. In the user list, select the administrator account you are logged in with and double-click on it.

4. Change the password on tab **General**.

2.5 Licenses and registrations

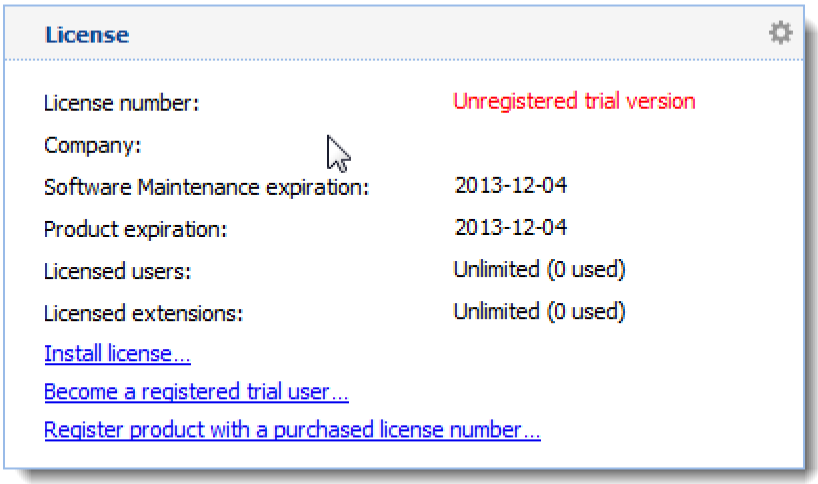
You can register the product from the welcome page of the administration interface which is displayed after each login.

> **WARNING**
> If Kerio Operator is protected by a firewall, it is necessary to allow outgoing HTTPS traffic for Kerio Operator at port 443. Unless HTTPS traffic is allowed, Kerio Operator cannot use the port to connect to the Kerio Technologies registration server.

When installed, the product can be registered as trial or as a full version.

2.5.1 Why should you register the trial version?

The trial version is intended to allow the customer to become familiar with the product's features and configuration. Once you register the trial version, you will be provided free Kerio Technologies technical support during the entire trial period (up to 30 days).



The trial version can be registered by clicking **Become a registered trial user** from the **Dashboard** (see the screenshot above). In the dialog box that appears, set the following parameters:

1. Enter security code (CAPTCHA) from the image.

2. Enter information about your company and agree with the privacy policy terms.

3. Choose how many computers do you have in your company and how you learned of Kerio Operator.

Now, a special identification code called Trial ID gets generated. This ID is later required for contacting the technical support. After a successful registration, Trial ID can be found in the license information in the administration interface.

> **NOTE**
> Once you purchase the product, your Trial ID will become your license number (it will not change).

2.5.2 Registering full version

If your trial version is registered, the license key (`licence.key` file) is automatically imported to your product within 24 hours from your purchase. The Trial ID you entered in your product upon registration will be activated as a standard license number.

If you haven't registered your trial version:

1. Open the administration interface.

2. Click **Register product with a purchased license number** on Dashboard.

3. In the first step of the registration, enter the license number and enter the security code from the image.

> **NOTE**
> The code is not case-sensitive.

4. Click **Next** to make Kerio Operator establish a connection to the registration server and check validity of the number entered. If the number is invalid, the registration cannot be completed.

5. Type the registration information about the company the product is registered to.

6. Kerio Operator connects to the registration server, checks whether the data inserted is correct and downloads automatically the license key (digital certificate).

7. Click **Finish** to close the wizard.

Manual import the license key

If you need to import a license key manually (for example from a backup), use the following steps:

1. Prepare the license key.
2. Log in to Kerio Operator administration.
3. Click **Install license** on **Dashboard**.
4. In the **Install License** dialog, click **Browse**.
5. In the **Open** dialog, find the file `.key` with the license key and click **Open**.
6. In the **Install License** dialog, click **OK**.
7. Check the result in the **License** tile on **Dashboard**.

Kerio Operator installs the licence key.

2.5.3 Registering via a web browser

You can also register Kerio Operator via web browser.

1. Go to https://secure.kerio.com/reg/
2. Register using your purchased license number.
3. By registering, you will receive a license key which must be imported to Kerio Operator.

> **NOTE**
> The trial version of Kerio Operator cannot be registered via the website.

2.5.4 How do I apply renewals or add-ons to my Kerio product?

When you purchase renewals or add-ons for a Kerio Product, License changes are applied automatically by the product within 24 hours.

You can also force an immediate update from the administration dashboard using the **update registration info**" link in the **License Details** tile.

2.6 Upgrading Kerio Operator

Choose your current Kerio Operator version for notes and instructions on how to upgrade to the latest version while retaining all settings:

- Kerio Operator 1.2.0 and newer
- Kerio Operator 1.1.3 and older

2.6.1 Manually uploading a binary image file

This procedure might be useful for the following situations:

- downgrade of Kerio Operator
- upgrade to a custom version (e.g. beta version)

If you have an upgrade image file, you can upload it manually:

1. In the administration interface, go to section **Advanced Options > tab Update Checker**.

2. Click the **Upload binary image** button.

3. Select the upgrade image file (`kerio-Operator-upgrade.img`).

4. Click the **Open** button. Wait for uploading the file.

5. Click the **Upgrade** button. Wait for the upgrade and restart of Kerio Operator.

When the restart is finished, your Kerio Operator is up-to-date.

2.6.2 Upgrading from versions 1.2.0 and newer

Learn how to upgrade Kerio Operator to the latest version while retaining all settings.

Important notes when upgrading

» An active and valid Software Maintenance is required to upgrade to new versions of Kerio Operator and its components as soon as they are available.

» Backup the Kerio Operator configuration before upgrade. For more information, refer to Saving Kerio Operator configuration to MyKerio (page 144).

» Check that the server meets the latest system and hardware requirements.

» Kerio Operator requires restarts during upgrade. Perform the upgrade when there is no traffic on the server or when it is least impacting on the business operation.

Upgrade procedure

1. From the administration console, go to **Advanced Options > Update Checker**.

2. Select the **Periodically check for new versions** option, so Kerio Operator checks for new updates every 24 hours.

3. If you want to download new versions automatically, select **Download new versions automatically**. If you want to get also beta versions of the product, select **Check also for beta versions**.

4. Click **Apply**.

5. When Kerio Operator finds a new version, click **Upgrade** to install it. Click **Yes** to confirm.

6. Kerio Operator then upgrades to the latest version and restarts automatically when done.

2.6.3 Upgrading from versions 1.1.3 and older

Learn how to upgrade Kerio Operator to the latest version while retaining all settings.

Important notes when upgrading

» An active and valid Software Maintenance is required to upgrade to new versions of Kerio Operator and its components as soon as they are available.

» Backup the Kerio Operator configuration before upgrade. For more information, refer to Saving Kerio Operator configuration to MyKerio (page 144).

» Check that the server meets the latest system and hardware requirements.

» Kerio Operator requires restarts during upgrade. Perform the upgrade when there is no traffic on the server or when it is least impacting on the business operation.

Upgrade procedure

1. Increase the file upload limit so Kerio Operator can be upgraded to a newer version. To do this, from the administration console, go to **Advanced Options** and set the maximum value in the **Maximum webserver upload file size** field.

2. Restart Kerio Operator.

3. Download Kerio Operator 1.2.0.

4. Go to **Advanced Options > Update Checker**, click the **Upload binary image** button, select the upgrade image file, wait until the file is uploaded and click **Upgrade**.

5. After Kerio Operator is restarted, go to **Advanced Options >Update Checker**, select the **Periodically check for new versions** option, and click **Apply**.

6. After Kerio Operator finds the latest version, click **Upgrade**, and click **Yes** to confirm the action. When the upgrade process is done, Kerio Operator restarts automatically.

2.7 Provider setup

This section helps you connect to various SIP providers.

2.7.1 Connecting to VoIP service providers

> **NOTE**
> This information is designed for Kerio Operator 2.4 and newer.s

You can connect Kerio Operator either to your VoIP service provider's SIP server or to a standard phone network. This topic discusses about connecting to a VoIP service provider.

Prerequisites

Before you configure an interface, you need the following information:

» Telephone number (or numbers) from your SIP provider.

» Domain/hostname of SIP server.

» Username and password for authentication.

» At least one internal extension defined in Kerio Operator — preferably the extension of an employee who redirects the calls.

Adding an interface

To configure an interface, you must first configure call routing. Once you configure incoming call routing, a configuration wizard configures outgoing call routing automatically.

1. In the administration interface, go to **Configuration > Call Routing** and click **Add SIP interface**. This displays the configuration wizard.

2. Key in a name for the interface (for example, the provider's name). The name must not contain spaces or special characters and must be unique.

3. Select **New provider**. The configuration differs for settings with one number or multiple numbers and for a SIP trunk with an interval of phone numbers.

One or multiple phone numbers

1. If you acquire one or multiple phone numbers from your provider, key in the numbers in the **New provider > With external number** field. You can:

- Separate individual numbers with commas (for example, `555450, 555451, 555452,` and so on).
- Key in a range using a dash (for example, `555450-555459`).

2. Click **Next**.

3. Select an extension that receives all calls from the provider.

4. Optionally, in the **Prefix to dial out** field, key in a prefix for outgoing calls and click **Next**. Kerio Operator uses this prefix to route calls to your provider's SIP server. This prefix can be the same for other providers. For more information, refer to Working with prefixes for outgoing calls (page 202).

5. Key in the domain name or the IP address acquired from your provider and if the server requires authentication, also key in the username and password.

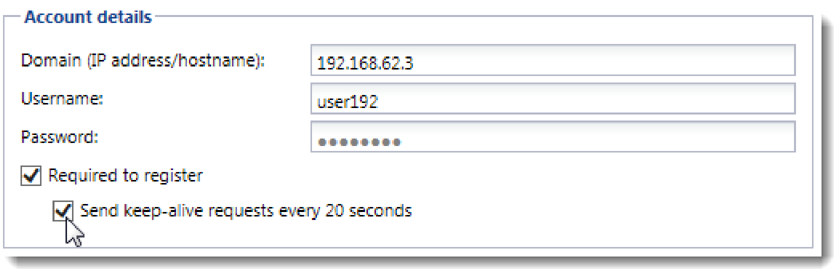
6. Select **Required to register** (the majority of providers require registration to a SIP server) and click **Next**.

7. Verify the information in the **Summary** section. If you need to add more information from your provider (for example, outbound proxy, inbound proxy, registrar, and so on), select the **Edit details of the created interface** option. For more information, refer to Configuring additional SIP details (page 27).

8. Click **Finish**.

9. Optionally, double-click the interface and enable the **Send keep-alive requests every 20 seconds** option.

> **WARNING**
> If your SIP provider does not send keep-alive packets, or your firewall or router has short and unchangeable NAT timeout for UDP connections, use this option to keep the UDP session open.



10. Click **OK** to save your changes.

11. Create a rewriting rule to correctly map numbers to internal user extensions. For more information, refer to Mapping external and internal numbers (page 196).

Interval of numbers

1. If you acquire a SIP trunk with an interval of phone numbers from your provider, key in `x` in place of the digits that vary (for example, `555xxx`).

2. Click **Next**.

3. Select the extension on which you want Kerio Operator to redirect all calls to unassigned (unused) extensions.

4. Optionally, in the **Prefix to dial out** field, key in a prefix for outgoing calls. Kerio Operator uses the prefix to route calls to your provider's SIP server. This prefix can be the same for other providers. For more information, refer to Working with prefixes for outgoing calls (page 202).

5. Click **Next**.

6. Key in the domain name or the IP address acquired from your provider. If the server requires authentication, also key in the username and password.

7. Select the **Required to register** option if the provider requires registration. With large number intervals, some providers do not require registration. Instead they use the IP address of your Kerio Operator. The address must be static and the provider needs to know about any changes that may occur.

8. Verify the information in the **Summary** section. If you need to add more information from your provider (for example, outbound proxy, inbound proxy, registrar, and so on), select the **Edit SIP details of created interface** option. For more information, refer to Configuring additional SIP details (page 27).

9. Click **Finish**.

10. Create a rewriting rule to correctly map numbers to internal user extensions. For more information, refer to Mapping external and internal numbers (page 196).

Configuring additional SIP details

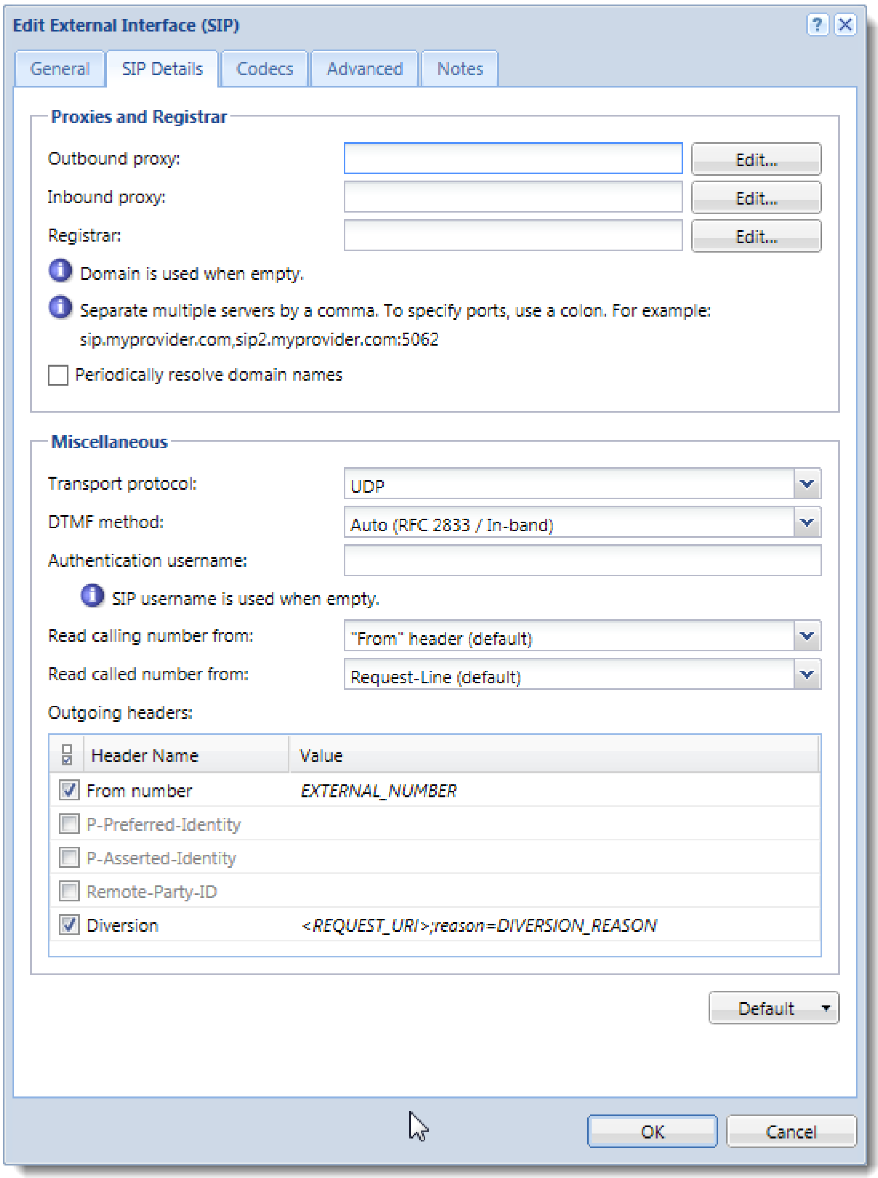
To set additional settings in your interface for incoming and outgoing calls:

1. In the administration interface, go to **Configuration > Call Routing > Interfaces and routing of incoming calls**, select a SIP interface and click **Edit**.

2. On the **SIP Details** tab, you can:

- Key in addresses to outbound proxy, inbound proxy and registrar (Kerio Operator uses domain by default).
- Change the transport protocol.
- Change the DTMF method.
- Key in an authentication username (Kerio Operator uses the SIP username by default).
- Change outgoing headers.

3. Click **OK** to save your changes.

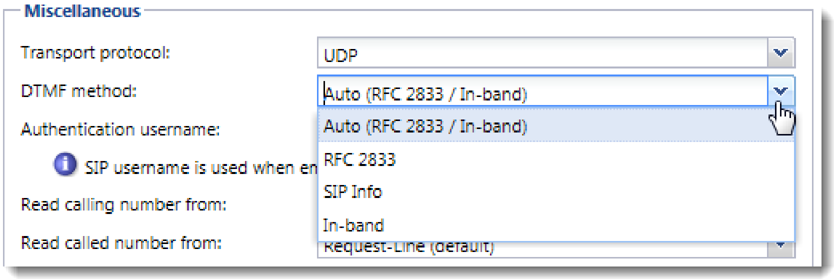


Configuring DTMF method

> **NOTE**
> This functionality exists since Kerio Operator 2.4.

For some SIP providers, the default configuration of DTMF detection, **Auto (RFC 2833 / In-band)**, does not work. You must find out the correct method from your SIP provider and configure it manually, as follows:

1. In the administration interface, go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.
2. Select a SIP interface and click **Edit**.
3. Go to the **SIP Details** tab.
4. Select the correct **DTMF method**.
5. Click **OK**



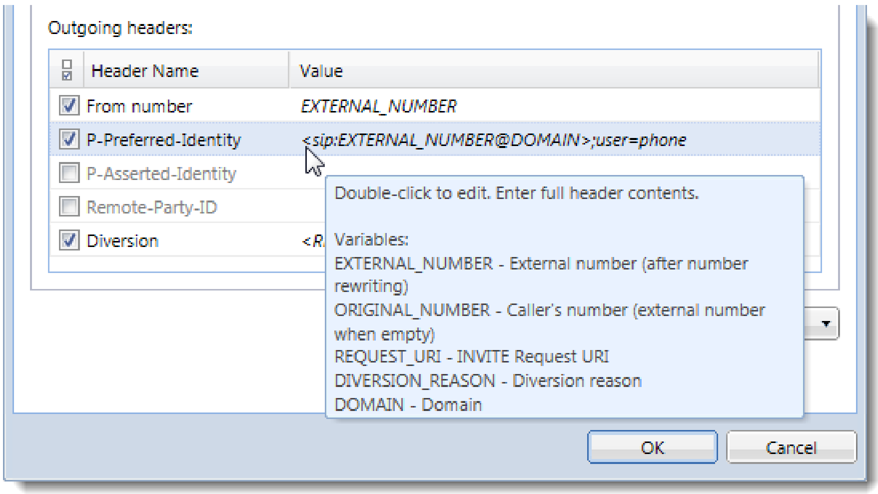
Configuring outgoing headers

> **NOTE**
> This functionality exists since Kerio Operator 2.4.

For some providers, you must add additional configuration to the SIP headers as provided to you by them.

To configure outgoing headers:

1. In the administration interface, go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.
2. Select a SIP interface and click **Edit**.
3. Go to the **SIP Details** tab.
4. Enable the outgoing header (see the list of supported headers below).
5. Double-click in the **Value** column and key in the header content (see the list of supported variables below).
6. Click **OK**



Kerio Operator supports these headers:

- From number
- P-Preferred-Identity
- P-Asserted-Identity
- Remote-Party-ID
- Diversion

To edit headers, use these variables:

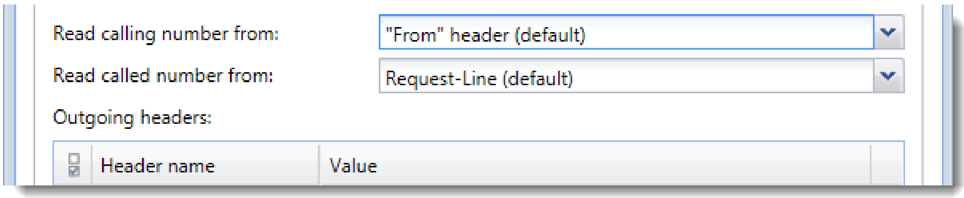
- `EXTERNAL_NUMBER` shows the external number after number rewriting
- `ORIGINAL_NUMBER` shows the number of the caller
- `REQUEST_URI` requests the information from the header of the forwarded call
- `DIVERSION_REASON` sends the reason of the call forwarding
- `DOMAIN` shows the domain of the interface

Reading the Caller ID from outgoing headers

> **NOTE**
> This functionality exists since Kerio Operator 2.4.4.

If your provider does not send the information about calling or called numbers in default headers (**From** for calling number and **Request-Line** for called numbers), you can configure Kerio Operator to read this information from different headers (for example, **P-Asserted-Identity**):

1. In the administration interface, go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.
2. Select a SIP interface and click **Edit**.
3. Go to the **SIP Details** tab.
4. For the fields **Read calling number from** and **Read called number from**, select a new header.
5. Click **OK** to save your settings.



Displaying the caller's number when transferring and redirecting calls

For more information, refer to Displaying the caller number when transferring and redirecting calls (page 31).

Resolving domain names of SIP providers

Your SIP providers may change their IP address for your registration without prior notice. To avoid inaccessibility, configure Kerio Operator to periodically resolve domain names and renew the registration:

1. In the administration interface, go to **Call Routing > Interfaces and routing of incoming calls**.
2. Select a SIP interface and click **Edit**.
3. Go to the **SIP Details** tab.
4. Select the **Periodically resolve domain names** option.
5. Click **OK**

Kerio Operator now periodically resolves domain names of your SIP provider and renews your registration whenever the IP address changes.

Mapping of numbers

For more information, refer to Mapping external and internal numbers (page 196).

2.7.2 Displaying the caller number when transferring and redirecting calls

Kerio Operator enables users to transfer or redirect their calls to another number or device. By default, the other device displays the number assigned to the extension from which the call is forwarded instead of the caller's number.

To solve this issue, enable additional outgoing headers to send the information about the call in them. Ask your provider which outgoing header to use.

Kerio Operator uses the diversion header by default.

> **NOTE**
> If your device can read this information, you might see, for example, a different icon or both numbers on your display.

> **NOTE**
> This configuration does not affect internal calls.

Configuring the diversion header

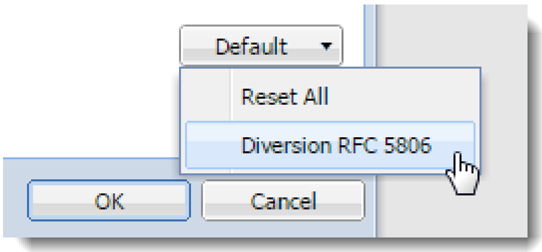
1. In the administration interface, go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.

2. Select a SIP interface and click **Edit**.

3. Go to the **SIP Details** tab.

4. Under the **Miscellaneous** section, open the drop down list and select **Diversion RFC 5806**. Kerio Operator automatically changes values in the From and Diversion headers.

5. Click **OK**



For more information, refer to Configuring outgoing headers (page 29).

Example

In this example:

- Meg Regret has the external number `377-338-9010`.
- John Smith has external number `555-1111`, which belongs to internal extension `11`.
- John configures call forwarding to his cell phone number `555-8888`.
- John wants to be able to return forwarded calls directly.
- The SIP provider uses a diversion header.

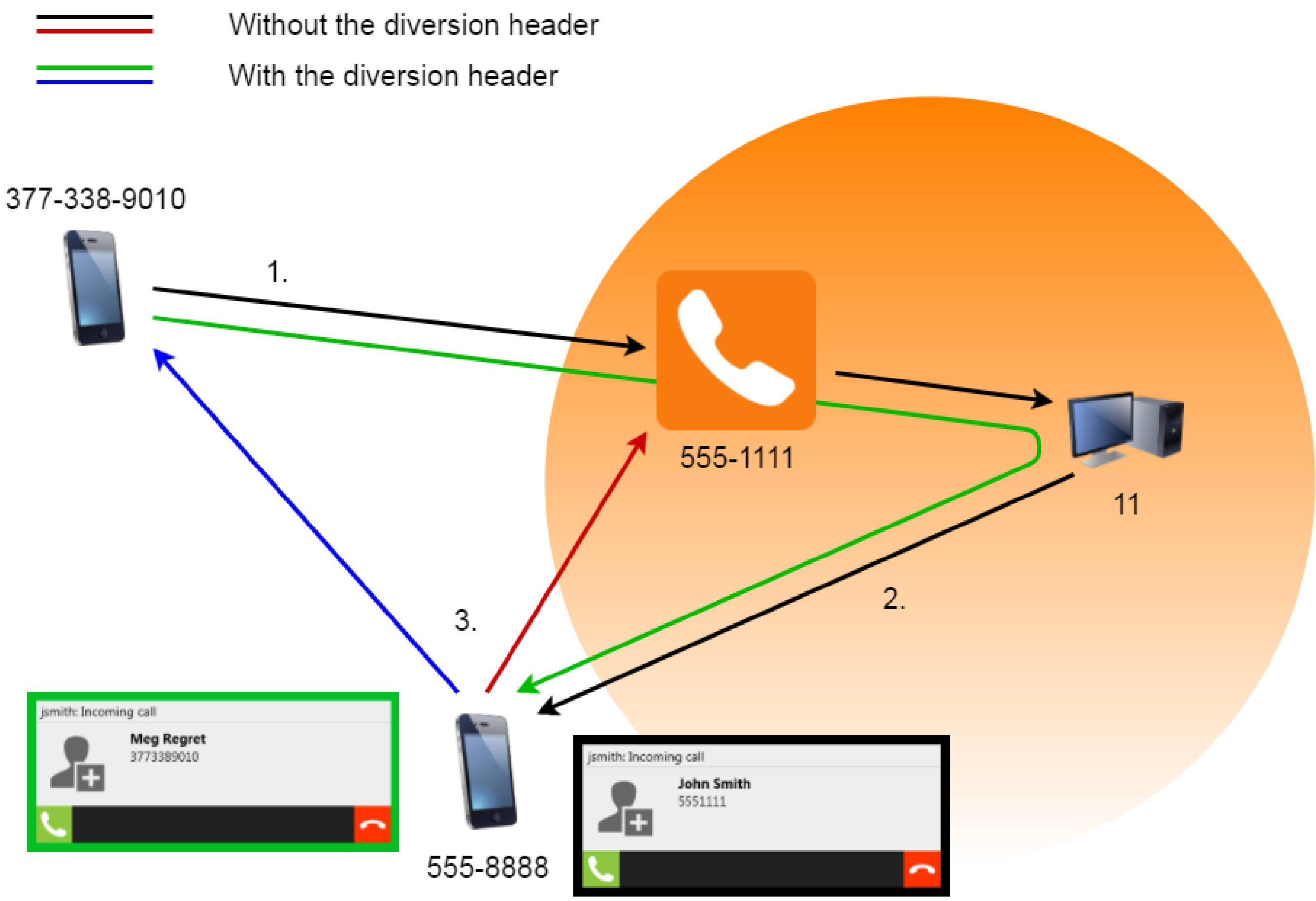
After Meg dials `555-1111`:

1. The SIP provider sends the call to Kerio Operator and the call reaches extension `11`.

2. Kerio Operator redirects the call to `555-8888`.

3. John Smith calls Meg Regret back. With the diversion header enabled, the call goes directly to Meg Regret. With the diversion header disabled, the call goes back to John's internal extension.

The difference is that, although the call takes the exact same path to the device, the diversion header allows John to read and see the caller's number, and dial back directly to that number.



2.7.3 Configuring Kerio Operator with NexVortex

You can configure Kerio Operator to send and receive calls using a SIP trunk from NexVortex.

Prerequisites

Before starting this procedure, you should have:

» The telephone number or numbers assigned to you by NexVortex. Each number will include the international country code without + at the beginning. For example, `14085555555`.

» Your SIP (PROXY) login credentials provided during the nexVortex account activation.

» If you have a firewall, make sure the SIP and RTP ports are properly routed to Kerio Operator. For more information, refer to Securing Kerio Operator (page 254).

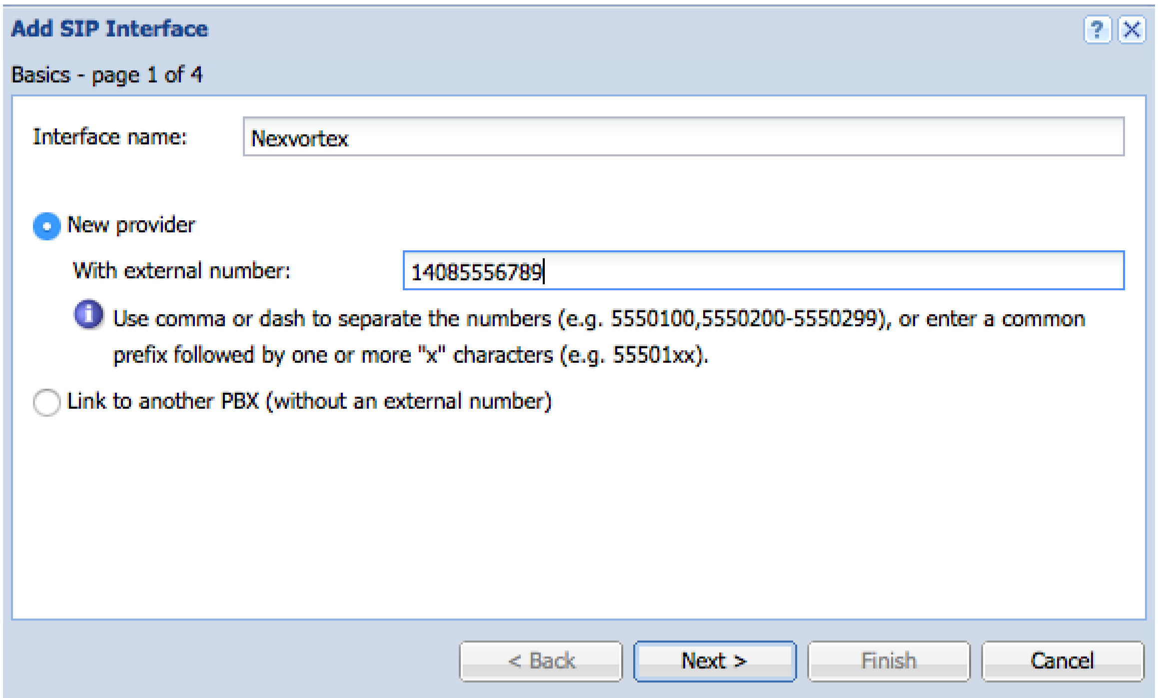
Configuration

1. Log in to the administration interface of Kerio Operator.

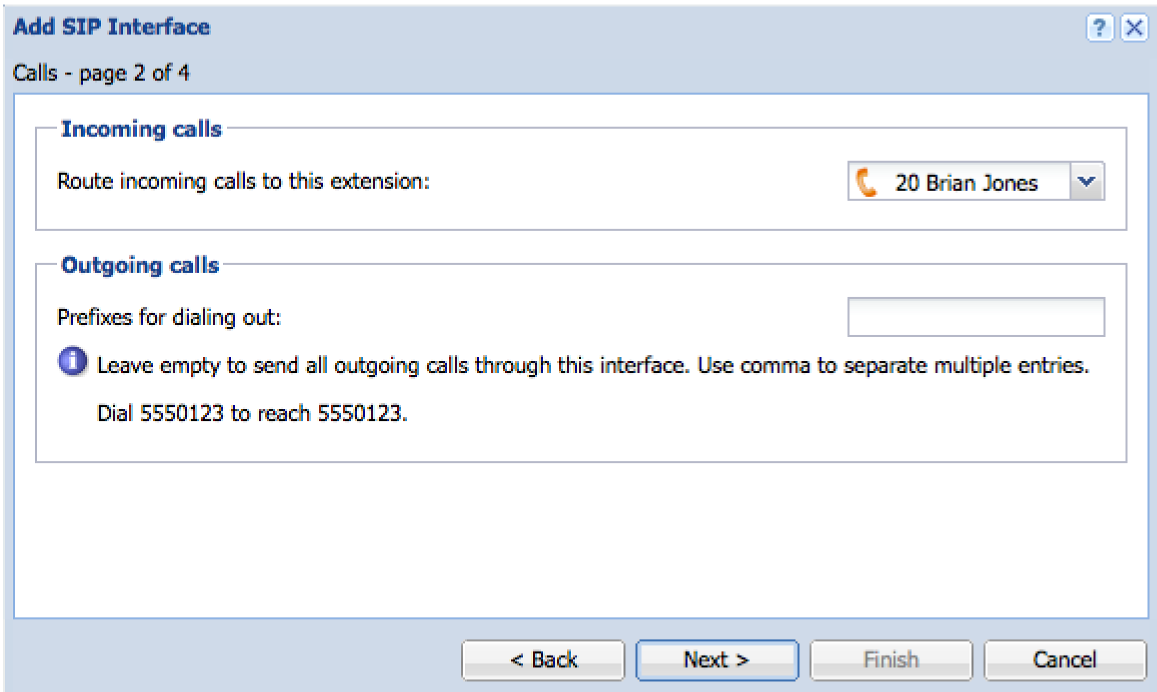
2. Go to the Call Routing screen

3. Click **Add a SIP Interface...**"

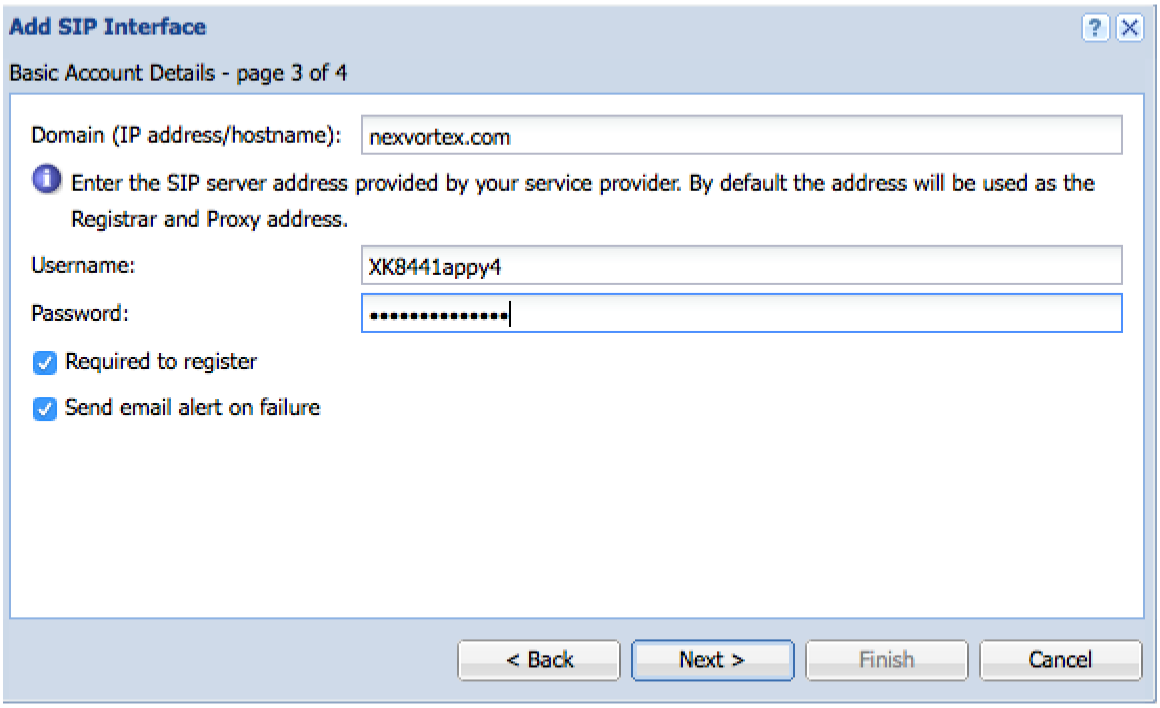
4. Enter an interface name (e.g. "nexvortex") and your external number or numbers.



5. Choose the extension to receive incoming calls



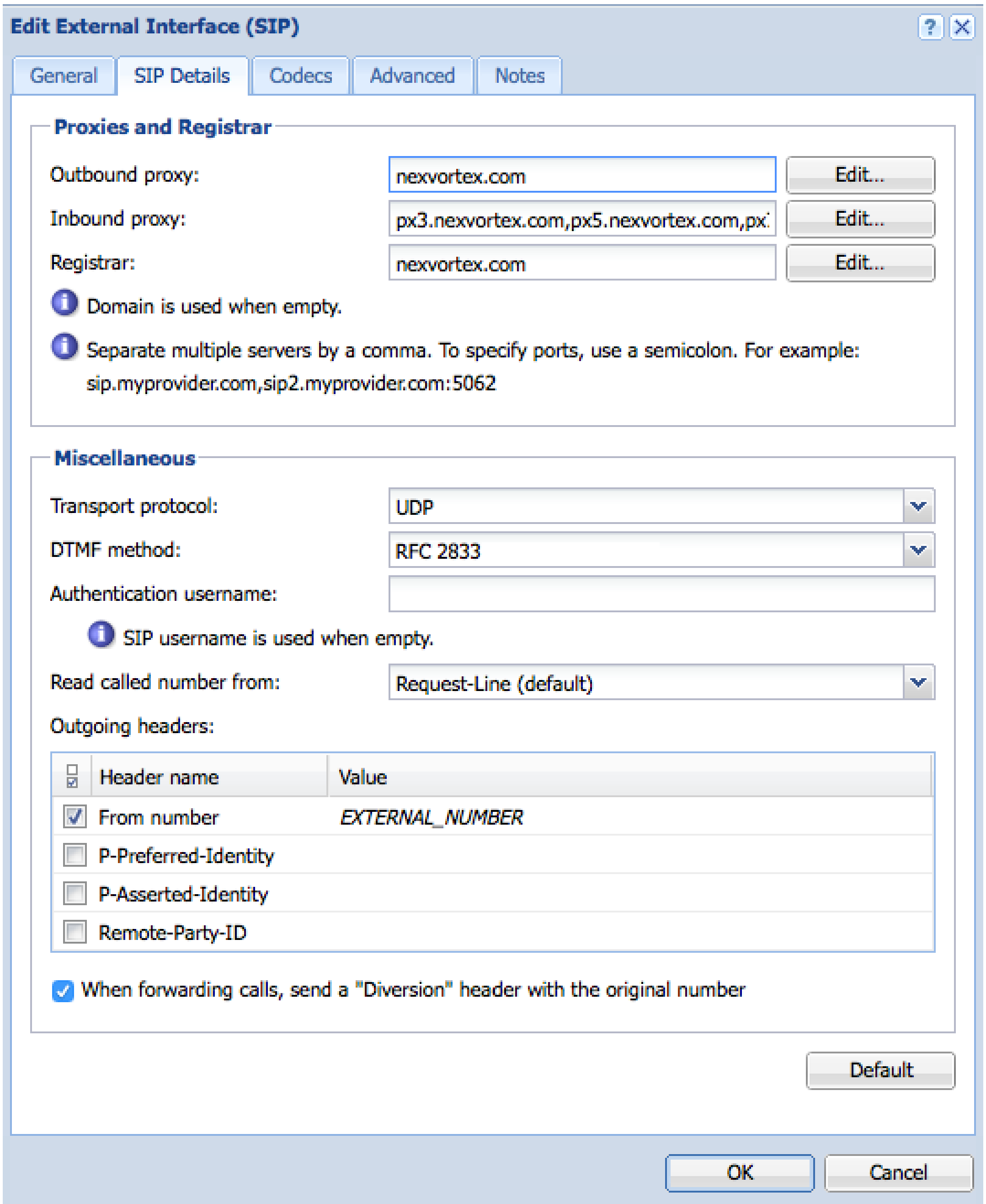
6. Specify the hostname (nexvortex.com) along with your Username and Password. Enable **Required to register**.



7. On the last page of the wizard, enable **Edit details of the created interface**.

8. In the SIP Details dialog configure the following:

- Outbound proxy: `nexvortex.com`
- Inbound proxy: `px3.nexvortex.com`,`px5.nexvortex.com`,`px7.nexvortex.com`
- Registrar: `nexvortex.com`
- DTMF method: `RFC 2833`
- Leave all other options with the default settings.



2.7.4 Connecting Kerio Operator to CenturyLink

> **NOTE**
> This topic is meant only for the CenturyLink IQ® SIP Trunk offer. CenturyLink supports only G.729 and G.711 U-law codecs.

Prerequisites

To connect your Kerio Operator to the CenturyLink provider, you need the following information:

» Your telephone numbers from CenturyLink. CenturyLink provides two types of numbers:

- **Trunk Pilot Number**, which is used as a SIP username.
- Trunk or range of external numbers, which are used for managing calls.

» The SIP username (**Trunk Pilot Number**) and password (**Trunk Group SIP Password**).

» The domain of CenturyLink (voip.centurylink.com).

» The outbound and inbound proxies of CenturyLink (**CenturyLink SBC IPv4 Address/Subnet Mask**).

» The registrar of CenturyLink (**CenturyLink SBC IPv4 Address/Subnet Mask**).

» The authentication username (**Trunk Group SIP ID**).

Configuration

1. In the administration interface of Kerio Operator, go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.

2. Click **Add SIP Interface**.

3. Key in a name for the interface.

4. In the **With external number** field, key in the trunk or range of numbers and click **Next**.

5. Select an extension to which you want Kerio Operator to redirect all calls to unassigned extensions.

6. (Optional) In the **Prefix to dial out** field, key in a prefix for outgoing calls.

7. Click **Next**.

8. In the **Domain (IP address/hostname)** field, key in `voip.centurylink.com`.

9. In the **Username** field, key in your **Trunk Pilot Number**.

10. In the **Password** field, key in your **Trunk Group SIP Password**.

11. Select the **Required to register** option and click **Next**.

12. Select **Edit details on the created interface** and click **Finish**.

After you finish the configuration, the **Edit External Interface (SIP)** dialog box opens:

1. In the **Proxies and Registar** section, key in your **CenturyLink SBC IPv4 Address/Subnet Mask** into **Outbound proxy**, **Inbound proxy**, and **Registrar** fields.

2. Go to the **Miscellaneous** section.

3. In the **Authentication username** field, key in your **Trunk Group SIP ID**.

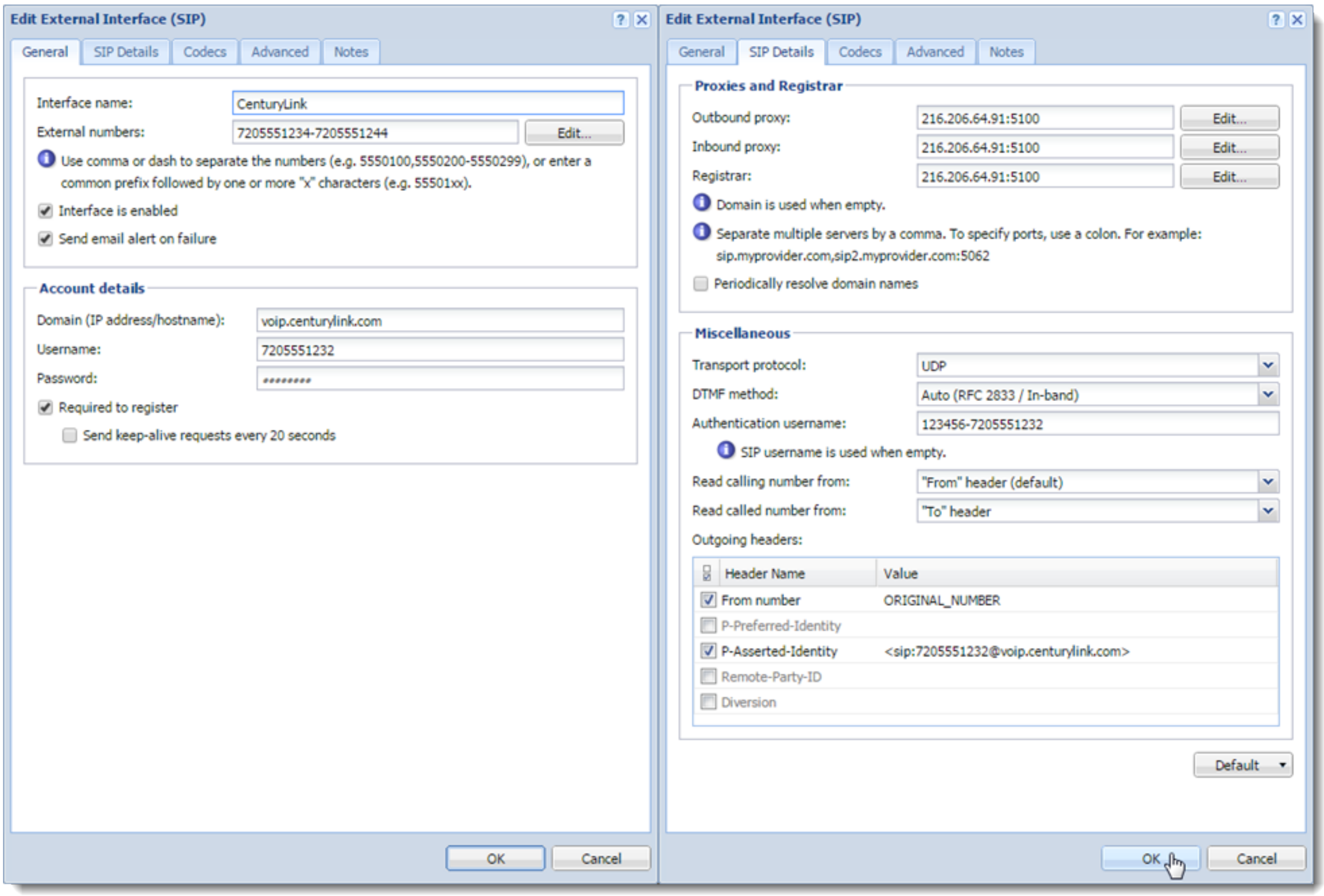
4. In the **Read called number from** field, select **"To" header**.

5. In the **Outgoing headers** table: Double-click the value for **From number** and key in `ORIGINAL_NUMBER`.

6. Enable the header.

7. Double-click the value for **P-Asserted-Identity** and key in `<sip:Trunk_Pilot_Number@voip.centurylink.com>`, where `Trunk_Pilot_Number` represents the number used for the SIP username.

8. Click **OK**



2.7.5 Connecting Kerio Operator to Deutsche Telekom

Prerequisites

To connect your Kerio Operator to Deutsche Telekom, you need the following information:

- Your telephone numbers from Deutsche Telekom
- The SIP username and the SIP password
- The domain of Deutsche Telekom. For example, `tel.t-online.de`.

Configuration

1. In the administration interface of Kerio Operator, go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.

2. Click **Add SIP Interface**.

3. Key in a name for the interface.

4. In the **With external number** field, key in your numbers.

> **NOTE**
> If you have multiple numbers from Deutsche Telekom, write individual numbers separated by a comma (for example, `555 5501, 555 5502, 555 5503, 555 5504`) or use a dash to define the range of numbers (for example, `555 5501—555 5504`).

5. Click **Next**.

6. Select an extension that receives all calls from the provider by default.

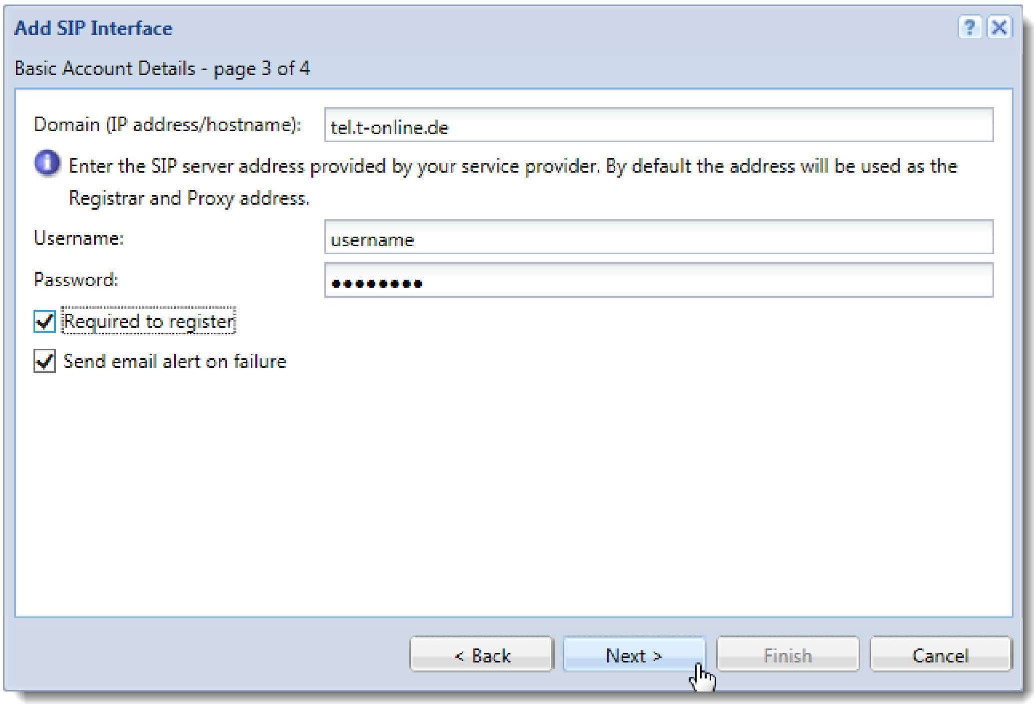
7. Optionally, In the **Prefix to dial out** field, you can key in a prefix for outgoing calls.

8. Click **Next**.

9. In the **Domain (IP address/hostname)** field, key in `tel.t-online.de`.

10. Key in the username and password.

11. Select the **Required to register** option.



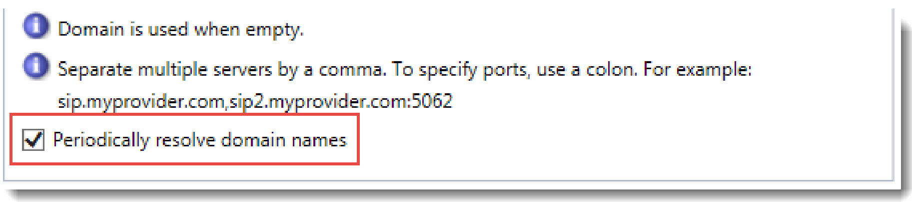
12. Click **Next**.

13. Select the **Edit details of the created interface** option and click **Finish**.

After you finish the configuration wizard, the **Edit External Interface (SIP)** dialog box opens:

1. In the **Proxies and Registrar** section, enable the **Periodically resolve domain names** option.

2. Click **OK**



2.7.6 Connecting Kerio Operator to Easybell

Prerequisites

To connect your Kerio Operator to Easybell, you need the following information:

» Your telephone numbers from Easybell.

» The SIP username and password.

» The domain of Easybell (`sip.easybell.de`).

Configuration

1. In the administration interface of Kerio Operator, go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.

2. Click **Add SIP Interface**.

3. Key in a name for the interface.

4. In the **With external number** field, key in the range of your numbers.

5. Click **Next**.

6. Select the extension to which you want Kerio Operator to redirect all calls to unassigned (unused) extensions.

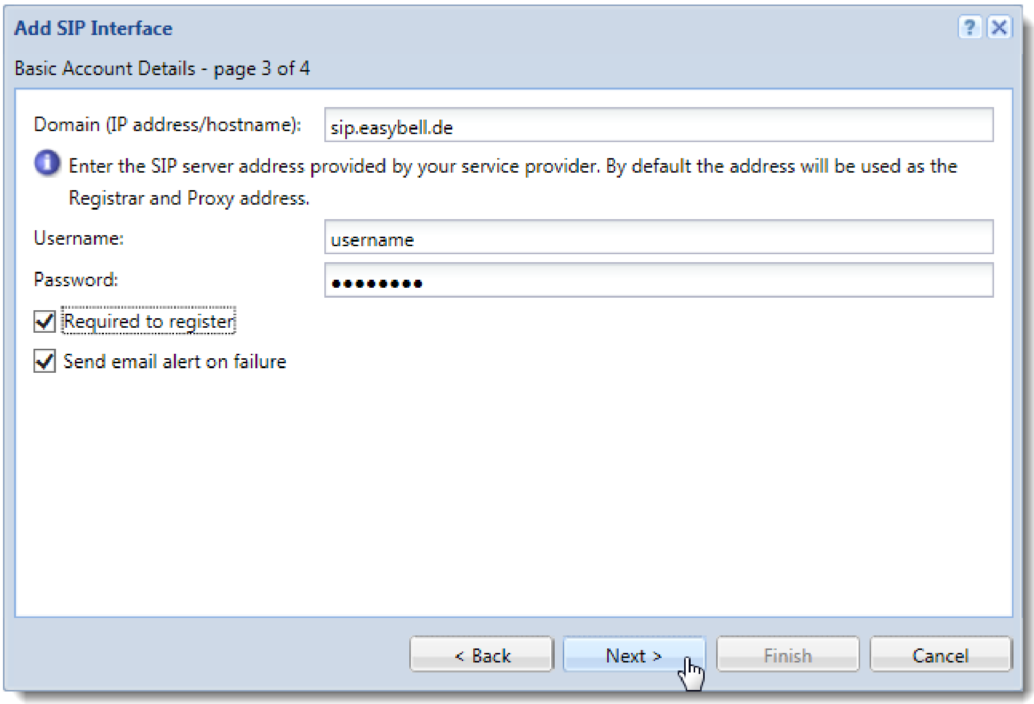
7. Optionally, in the **Prefix to dial out** field, you can key in a prefix for outgoing calls.

8. Click **Next**.

9. In the **Domain (IP address/hostname)** field, key in `sip.easybell.de`.

10. Key in the username and password.

11. Select the **Required to register** option.

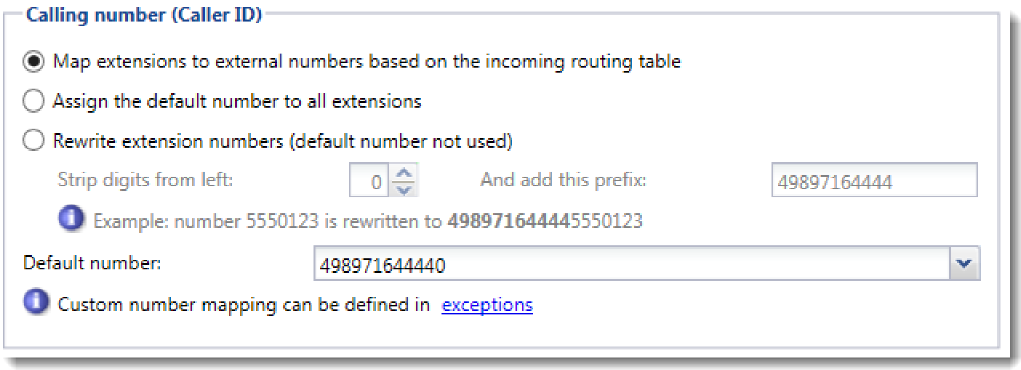


12. Click **Next**.

13. Verify the information in the **Summary** section and click **Finish**.

14. Double-click the created outgoing route for this interface.

15. In the **Calling number (Caller ID)** section, select **Map extensions to external numbers based on the incoming routing table** and click **OK**.



2.7.7 Connecting Kerio Operator to Net2Phone

Learn how to configure Kerio Operator with a SIP Trunk to Net2Phone.

> **NOTE**
> This information is designed for Kerio Operator 2.4 and newer.

Prerequisites

Before you start the configuration, you need the following information:

» The telephone number or numbers assigned to you by Net2Phone. Each number will include the US international country code (without + at the beginning). For example, `14085555555`.

» Your SIP (PROXY) login credentials provided during the Net2Phone account activation.

Configuration

1. In the administration interface, go to **Configuration > Call Routing**.

2. Click **Add SIP Interface**.

3. In the first screen, key in an interface name (for example, `Net2Phone`), select **New provider** and key in your telephone number. For example, `14085555555`. In case of multiple numbers, use comma separation as noted in the dialog.

4. Click **Next**.

5. Choose the extension to receive incoming calls and key in a prefix that will be used for external calls (for example, 9).

6. Click **Next**.

7. In the **Domain (IP address/hostname)** field, specify the hostname and credentials as provided by Net2Phone (`ippbx.net2phone.com`).

8. Check **Require to register**.

Check the settings by dialing an external phone number.

2.7.8 Connecting Kerio Operator to NEXCO Networks

> **NOTE**
> This information is designed for Kerio Operator 2.3.5 and older. For more information, refer to Connecting to VoIP service providers (page 25).

You can configure a SIP trunk with NEXCO Networks for dialing to the public telephone network. This topic describes the necessary configuration in Kerio Operator.

Prerequisites

After setting up an account with NEXCO Networks, you should be given the following information from the provider:

- The telephone number (or numbers) assigned to you.
- Trunk # (sometimes referred to as User Name or User ID).
- Password
- Server IP or Domain. For example, `media1.nexconetworks.net`.

This information is required in the configuration as described below.

Configuration

1. Log in to the web administration interface of Kerio Operator.

2. Go to **Configuration > Call Routing**.

3. Click **Add a SIP Interface.**

4. Enter an interface name (e.g. "NEXCO").

5. Choose **New provider** and enter your telephone number (you may need to add a `1` at the beginning of the telephone number). Use a comma to separate multiple phone numbers.

6. Click **Next.**

7. Choose the extension to receive incoming calls.

8. Enter a dial out prefix if necessary.

9. Click **Next.**

10. Specify the hostname provided by NEXCO Networks (**media1.nexconetworks.net**) and the default port **5060.**

11. Specify the **Username** (Trunk #) and **Password** (Password) values as provided by NEXCO Networks.

12. Enable **User ID differs from the telephone number** and enter your NEXCO Networks Trunk # (the same number that is in the Username field).

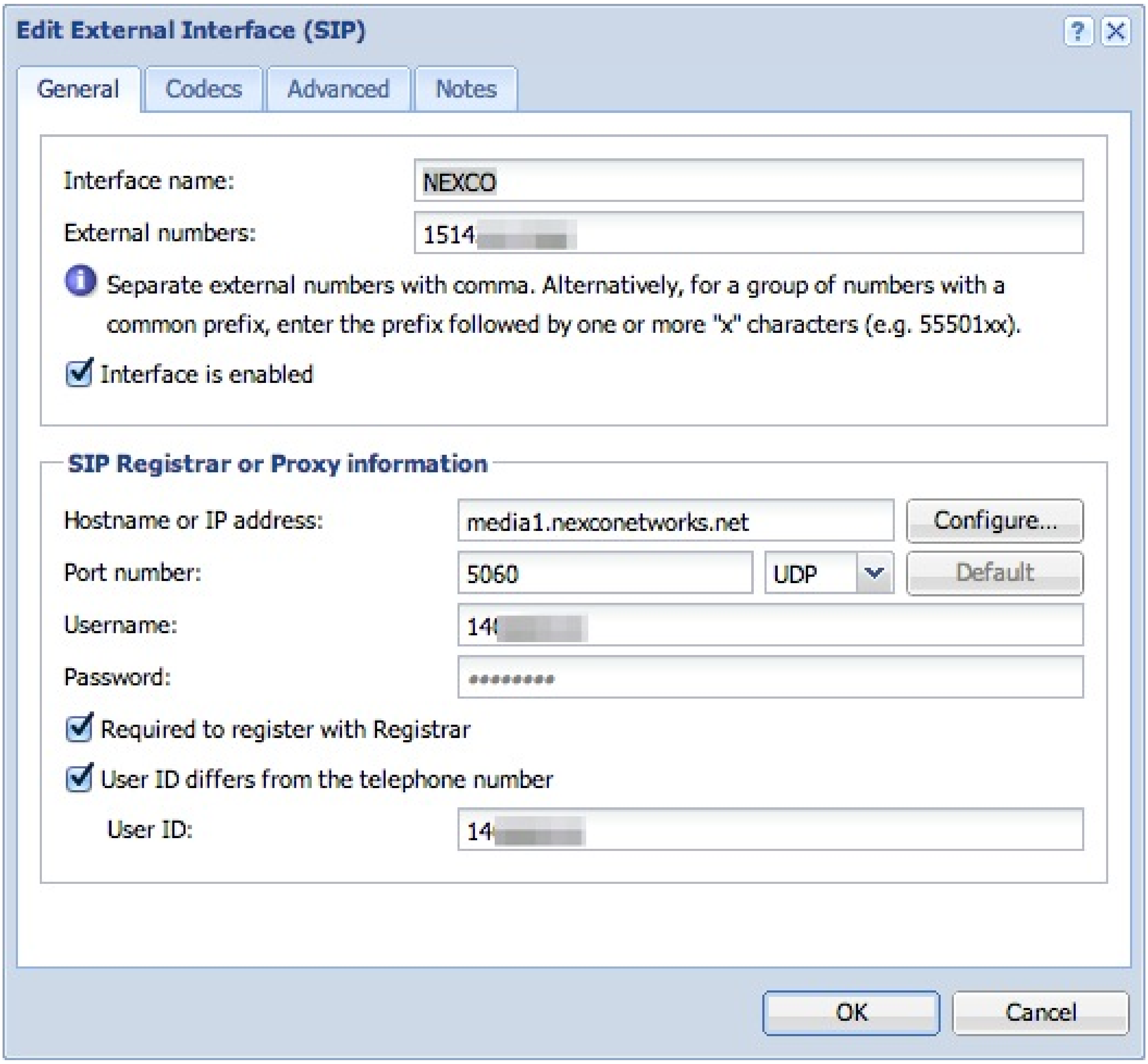
13. Enable the option **Register with registrar.**

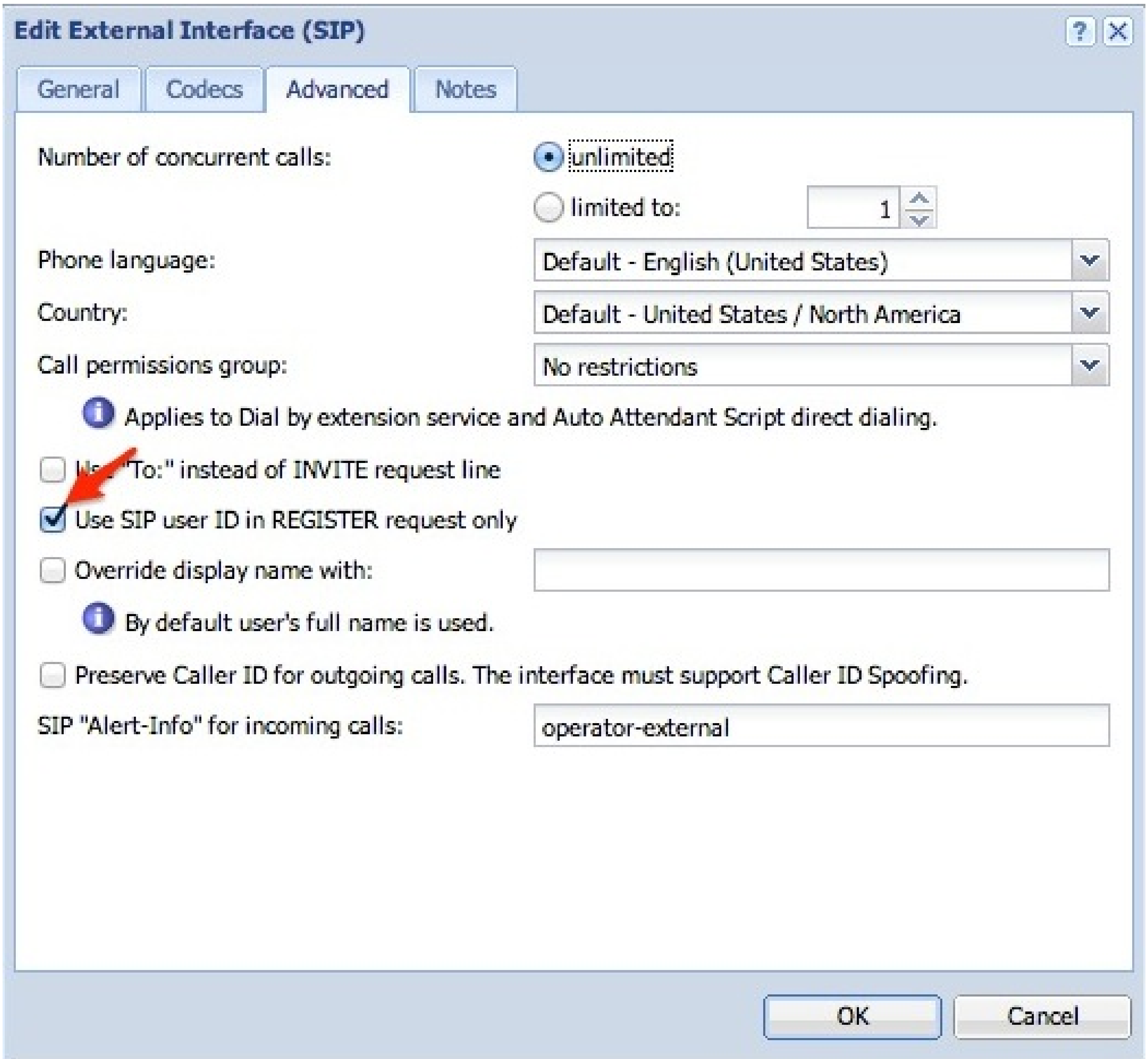
14. Click **Finish.**

15. In order to transmit the correct CallerID information on outbound calls you must edit the interface that you've just created by doing the following:

    - Select the interface you just created (e.g. NEXCO) and click **Edit.**
    - Click the **Advanced** tab.

- Check the box **Use SIP user ID in REGISTER request only.**
- Click **OK** to save.


Edit External Interface (SIP)
General
Codecs
Advanced
Notes
Interface name:
NEXCO
External numbers:
1514
Separate external numbers with comma. Alternatively, for a group of numbers with a common prefix, enter the prefix followed by one or more "x" characters (e.g. 55501xx).
Interface is enabled
SIP Registrar or Proxy information
Hostname or IP address:
media1.nexconetworks.net
Configure...
Port number:
5060
UDP
Default
Username:
14
Password:
••••••••
Required to register with Registrar
User ID differs from the telephone number
User ID:
14
OK
Cancel



2.7.9 Connecting Kerio Operator to QSC

Prerequisites

To connect your Kerio Operator to QSC, you need the following information:

- Your telephone numbers from QSC.
- The SIP username and the SIP password.
- The domain of QSC (sip.qsc.de).

Configuration

1. In the administration interface of Kerio Operator, go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.

2. Click **Add SIP Interface**.

3. Key in a name for the interface.

4. In the **With external number** field, key in the range of your numbers.

5. Click **Next**.

6. Select the extension to which you want Kerio Operator to redirect all calls to unassigned (unused) extensions.

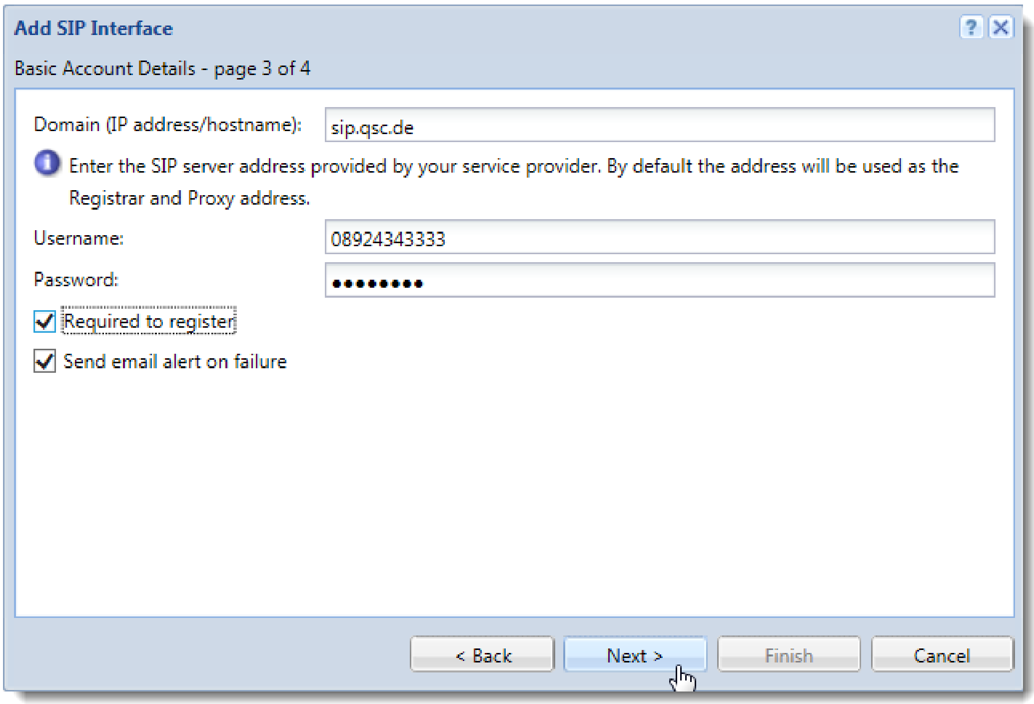
7. Optionally, in the **Prefix to dial out** field, you can key in a prefix for outgoing calls.

8. Click **Next**.

9. In the **Domain (IP address/hostname)** field, key in `sip.qsc.de`.

10. Key in the username and password.

11. Select the **Required to register** option.



12. Click **Next**.

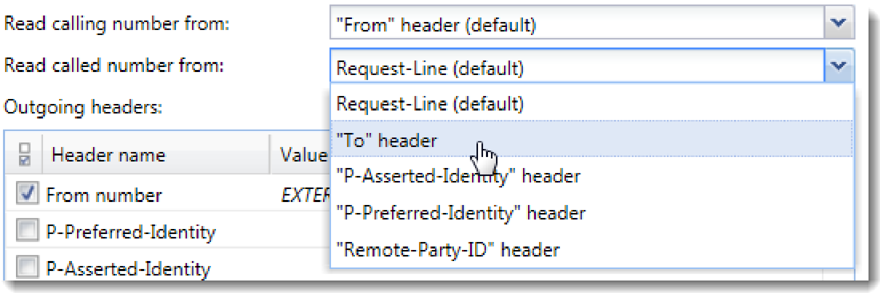
13. Select the **Edit details of the created interface** option and click **Finish**.

After you finish the configuration wizard, the **Edit External Interface (SIP)** dialog box opens:

1. Go to **SIP Details > Miscellaneous**.

2. In the **Read called number from** field, select the **"To" header** option.

3. Click **OK**



2.7.10 Connecting Kerio Operator to Sipgate.co.uk

Prerequisites

To connect your Kerio Operator to Sipgate.co.uk, you need the following information:

» Your phone numbers from Sipgate.co.uk

> **WARNING**
> In the Sipgate.co.uk account, Sipgate.co.uk displays numbers in the UK format. When you configure the numbers in Kerio Operator, you need to change the prefix **0** to a prefix **44**.
> For example, if you have numbers `056 0001 2345` and `056 0001 2346` from Sipgate.co.uk. When you configure the SIP interface in Kerio Operator, you change the prefix and key in numbers `4456 0001 2345` and `4456 0001 2346`.

» The SIP-ID of your SIP account. For example, `123456t0`.

» The SIP password of your SIP account.

» The Registry/Proxy address of your account. For example, `sipconnect.sipgate.co.uk`.

Configuration

In the administration interface of Kerio Operator:

1. Go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.

2. Click **Add SIP Interface**.

3. Key in a name for the interface.

4. In the **With external number** field, key in your numbers with changed prefixes. For more information, refer to Prerequisites (page 46)..

5. Click **Next**.

6. Select the extension that receives all calls from the provider.

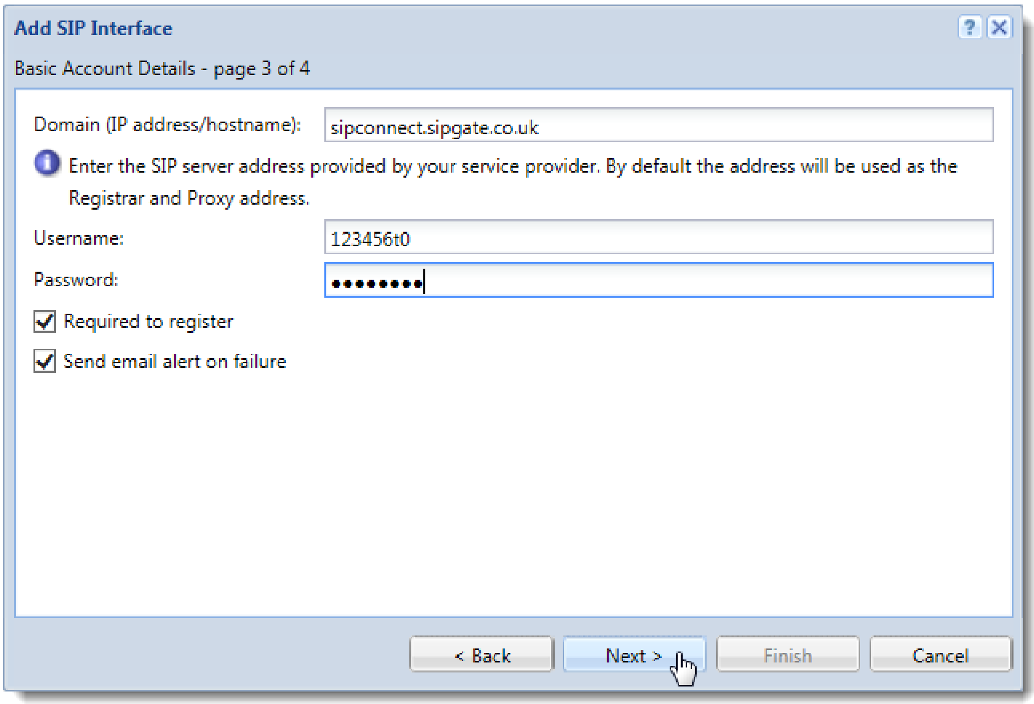
7. Optionally, in the **Prefix to dial out** field, you can key in a prefix for outgoing calls.

8. Click **Next**.

9. In the **Domain (IP address/hostname)** field, key in `sipconnect.sipgate.co.uk`.

10. Key in the username (`123456t0`) and the password.

11. Select the **Required to register** option and click **Next**.



12. Select the **Edit details of the created interface** option and click **Finish**.

Kerio Operator finishes the configuration wizard and the **Edit External Interface (SIP)** dialog box opens:

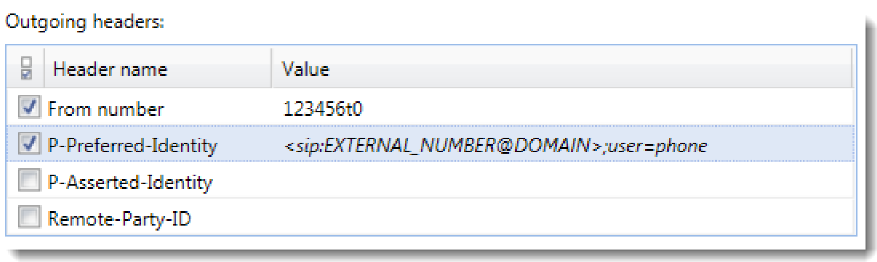
1. Go to the **Outgoing headers** table.

2. In the **From number** field, key in your SIP-ID (`123456t0`)

3. Optionally, select **P-Preferred Identity** and do not change the default value. If you have more than one phone number from Sipgate.co.uk, use this option to display the external numbers configured in Kerio Operator for your outgoing calls instead of the **Fallback Caller ID** number configured in your Sipgate account.

4. Click **OK**

You can now make some test calls to verify the connection to Sipgate.co.uk.



2.7.11 Connecting Kerio Operator to Sipgate Deutschland

Prerequisites

To connect your Kerio Operator to Sipgate Deutschland, you need the following information:

- Your telephone numbers from Sipgate Deutschland.

NOTE
You need to add the country code 49 to the external number.

» The SIP username and the SIP password.

» The domain of Sipgate Deutschland (sipconnect.sipgate.de).

Configuration

1. In the administration interface of Kerio Operator, go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.

2. Click **Add SIP Interface**.

3. Key in a name for the interface.

4. In the **With external number** field, key in the range of your numbers.

5. Click **Next**.

6. Select the extension to which you want Kerio Operator to redirect all calls to unassigned (unused) extensions.

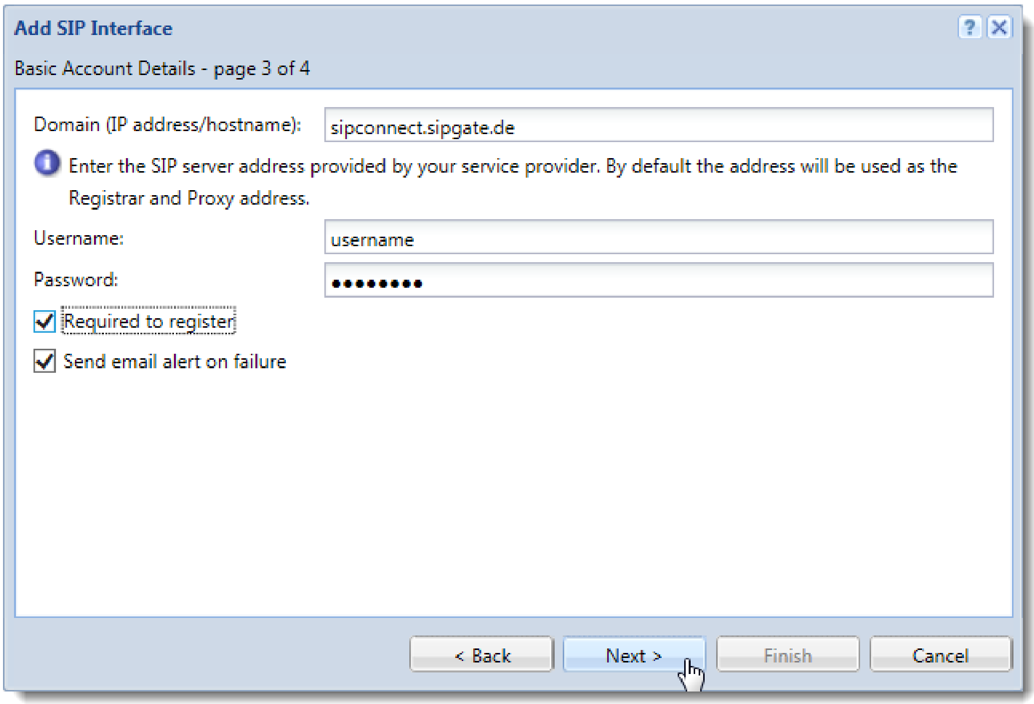
7. Optionally, in the **Prefix to dial out** field, you can key in a prefix for outgoing calls.

8. Click **Next**.

9. In the **Domain (IP address/hostname)** field, key in `sipconnect.sipgate.de`.

10. Key in the username and password.

11. Select the **Required to register** option.



12. Click **Next**.

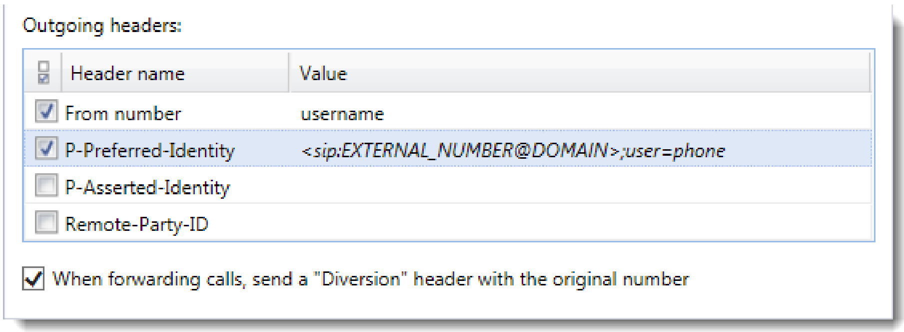
13. Select the **Edit details of the created interface** option and click **Finish**.

After you finish the configuration wizard, the **Edit External Interface (SIP)** dialog box opens:

1. Go to **SIP Details > Miscellaneous > Outgoing headers**.

2. In the **From number** header, key in your SIP username.

3. Select the **P-Preferred-Identity** header and click **OK**.



2.7.12 Connecting Kerio Operator to SIP.US and SIPTRUNK.COM

You can configure a SIP trunk with SIP.US or SIPTRUNK.COM for dialing to the public telephone network. This topic describes the necessary configuration in Kerio Operator.

Prerequisites

Accounts you setup in SIP.US or SIPTRUNK.COM include the following information (available in the SIP.US or SIPTRUNK.COM control panel) which is required during the configuration with Kerio Operator:

- The telephone number or numbers assigned to you.
- Trunk #
- Password

Configuration

1. Log in to the web administration interface of Kerio Operator.

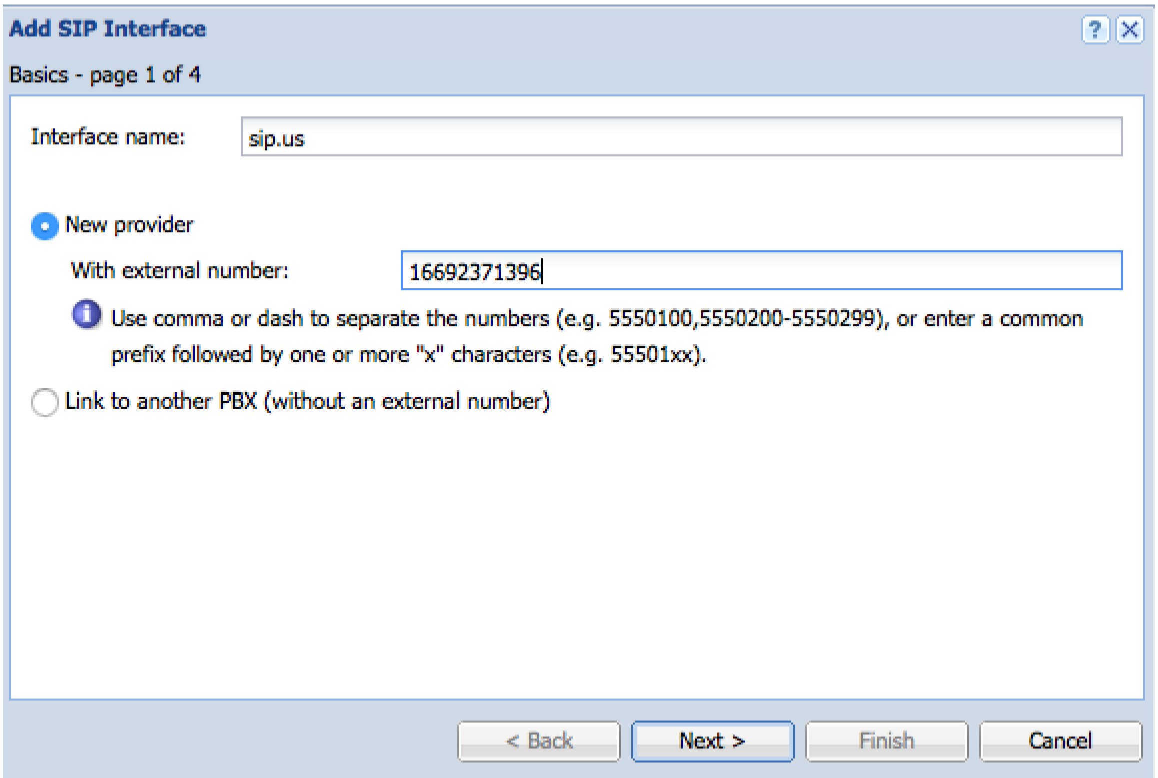
2. Go to **Configuration > Call Routing**.

3. Click **Add a SIP Interface**.

4. Assign an interface name.

5. Choose **New provider** and enter your telephone number (use a comma to separate multiple phone numbers).

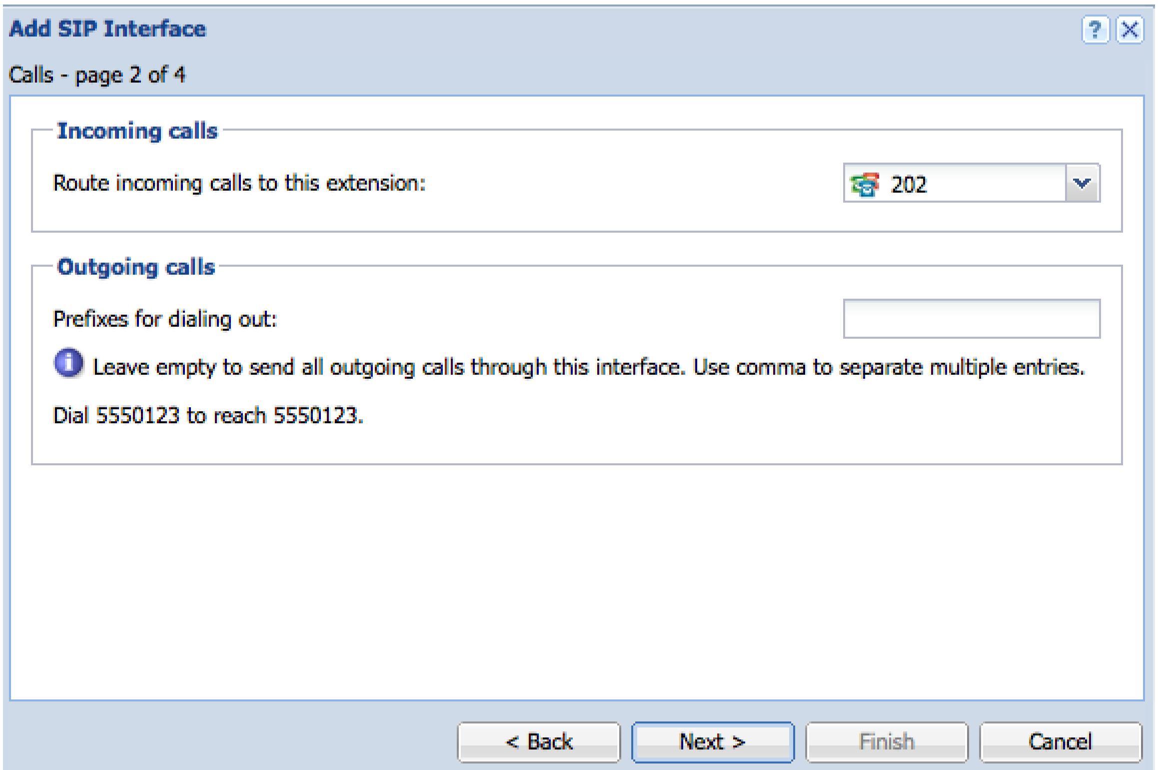
6. Click **Next**.



7. Choose the extension to receive incoming calls.

8. Enter a dial out prefix if necessary.

9. Click **Next**.

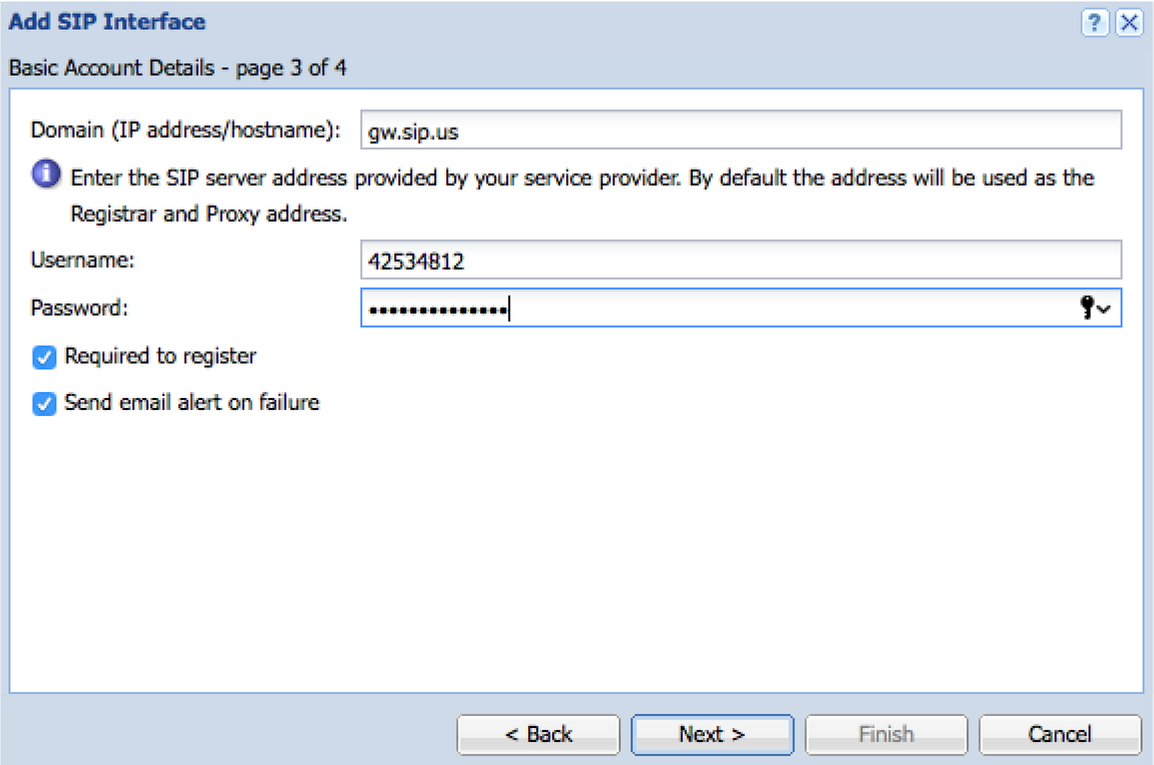


10. Specify the domain (gw.sip.us for SIP.US) or (gw.siptrunk.com for SIPTRUNK.COM)

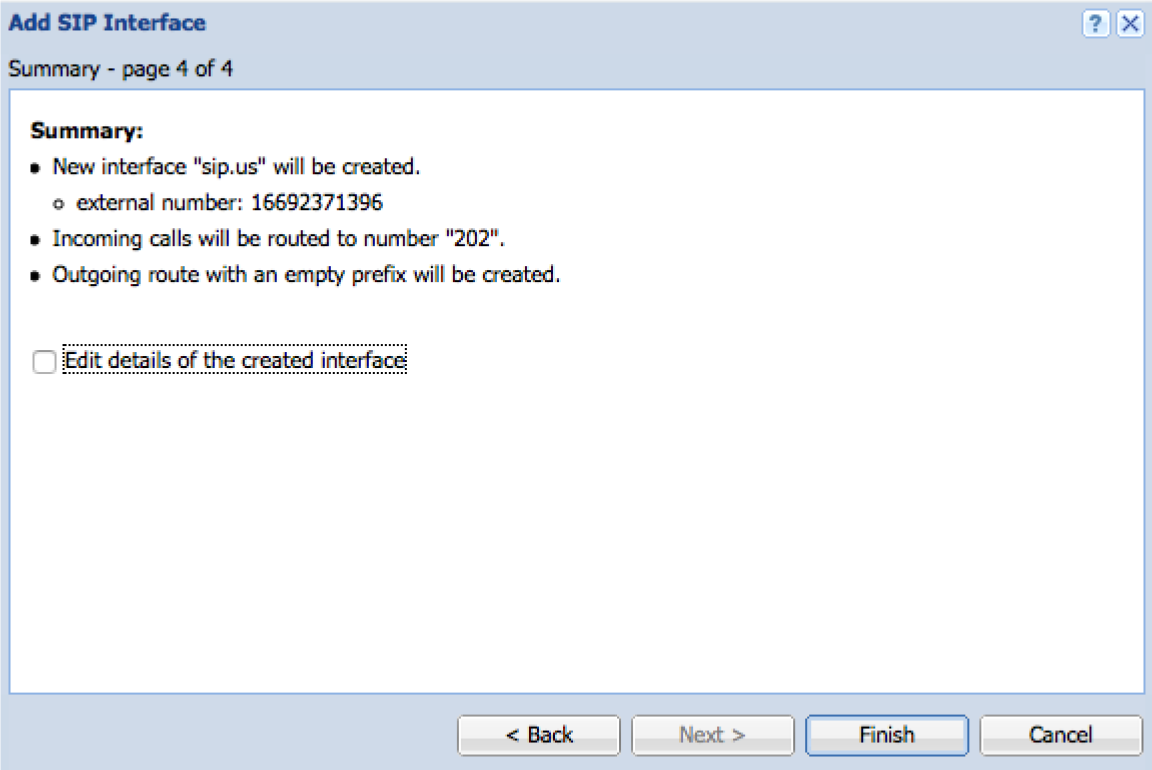
11. Specify the **Username** (Trunk #) and **Password** (Password) values.

12. Enable **Required to register**.

13. Click **Next**.



14. Review your settings and click **Finish**.



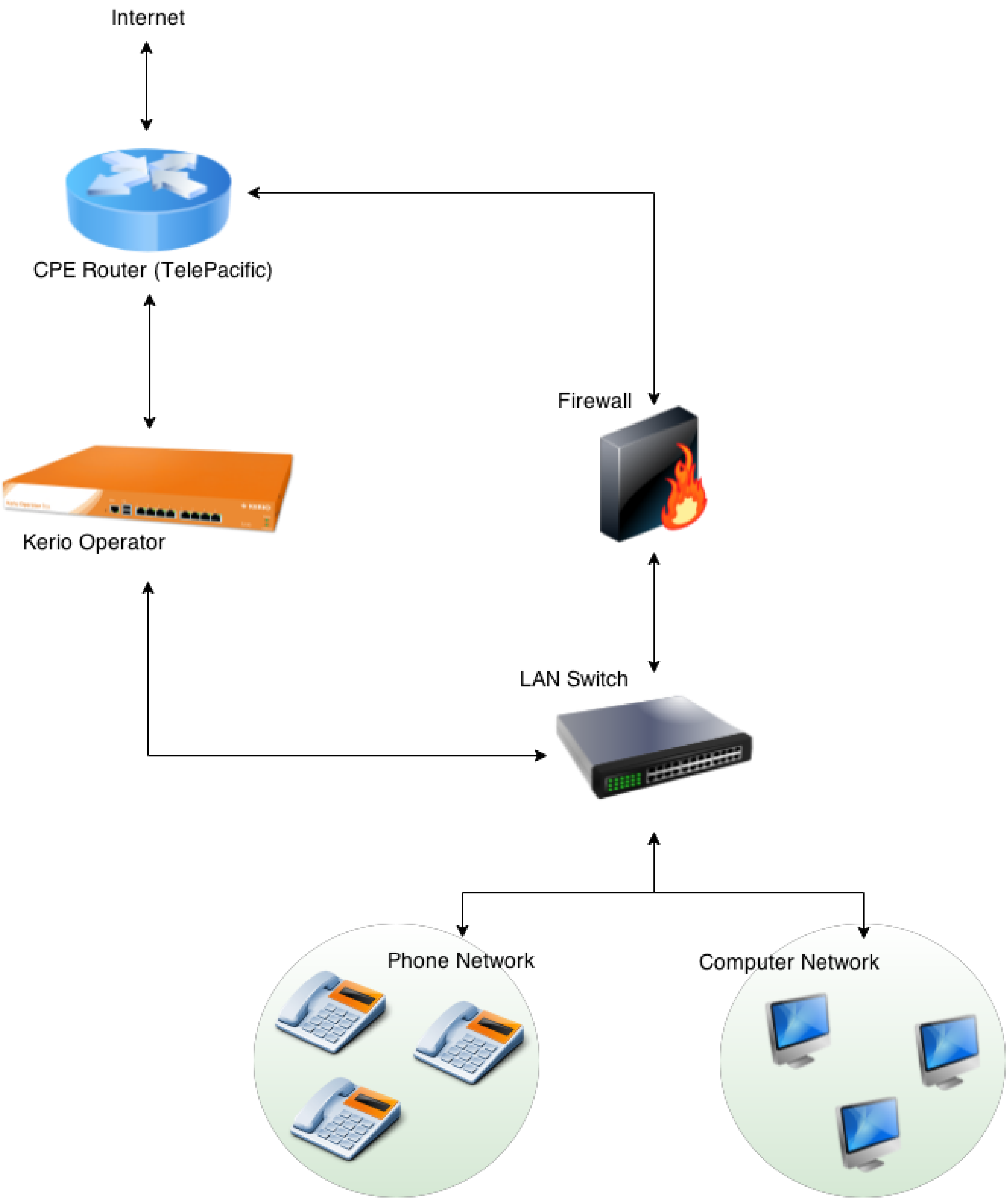
2.7.13 Connecting Kerio Operator to TelePacific

You can configure a SIP trunk with Telepacific for dialing to the public telephone network. This topic describes the necessary configuration in Kerio Operator.

Prerequisites

» To maximize call quality, TelePacific installs a customer-premises equipment (CPE) router at your physical location. It is necessary to properly design your network to support this type of configuration.

» Kerio Operator requires at least two network interfaces. One interface connects directly to the CPE router, and the other interface connects to your local area network. The diagram below illustrates the basic structure.



Configuring TCP/IP parameters in Kerio Operator

Configure the interface connecting to the CPE Router with a private IP address that you assign manually (e.g., 192.168.10.11/24). Do not assign a gateway to this interface. Configure the interface connecting to your local network with a static, or dynamic IP address that your DHCP server assigns to Kerio Operator. Use this interface to connect to the internet via the local area network.

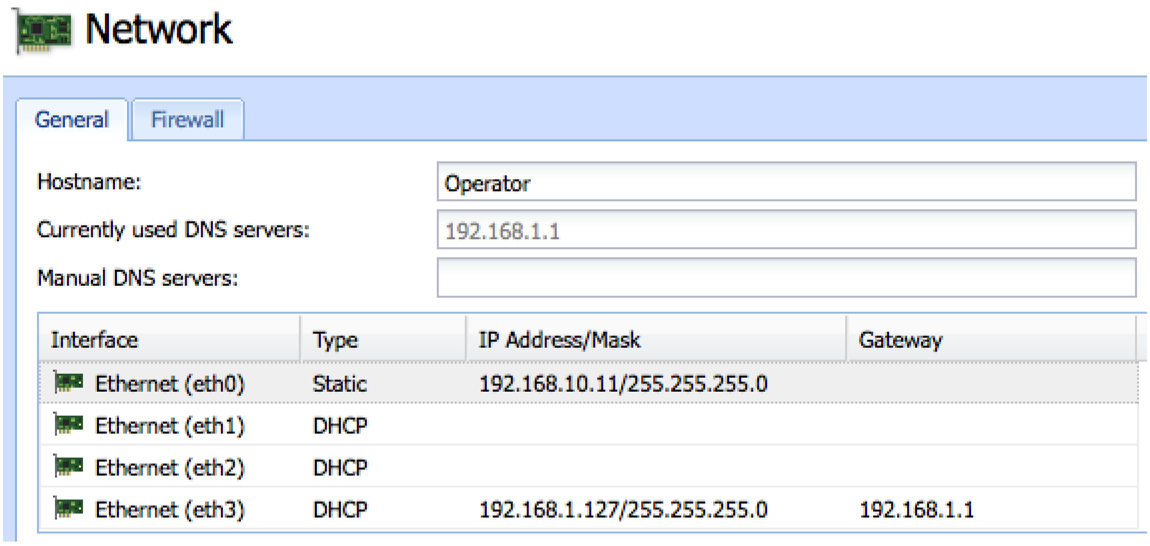
1. Log in to the administration interface

2. Go to **Configuration > Network**

3. **Edit** the interface connecting to the CPE router and assign the TCP/IP parameters for this private network

4. **Edit** the interface connecting to the local network and verify that the TCP/IP parameters are valid for the local area network

5. Click **Apply**



Configuration

1. Log in to the administration interface and go to **Configuration > Call Routing**

2. Click **Add a SIP Interface** and enter an interface name (e.g. "Telepacific")

3. Choose **New provider** and enter your telephone number (use a comma to separate multiple phone numbers)

4. Click **Next**

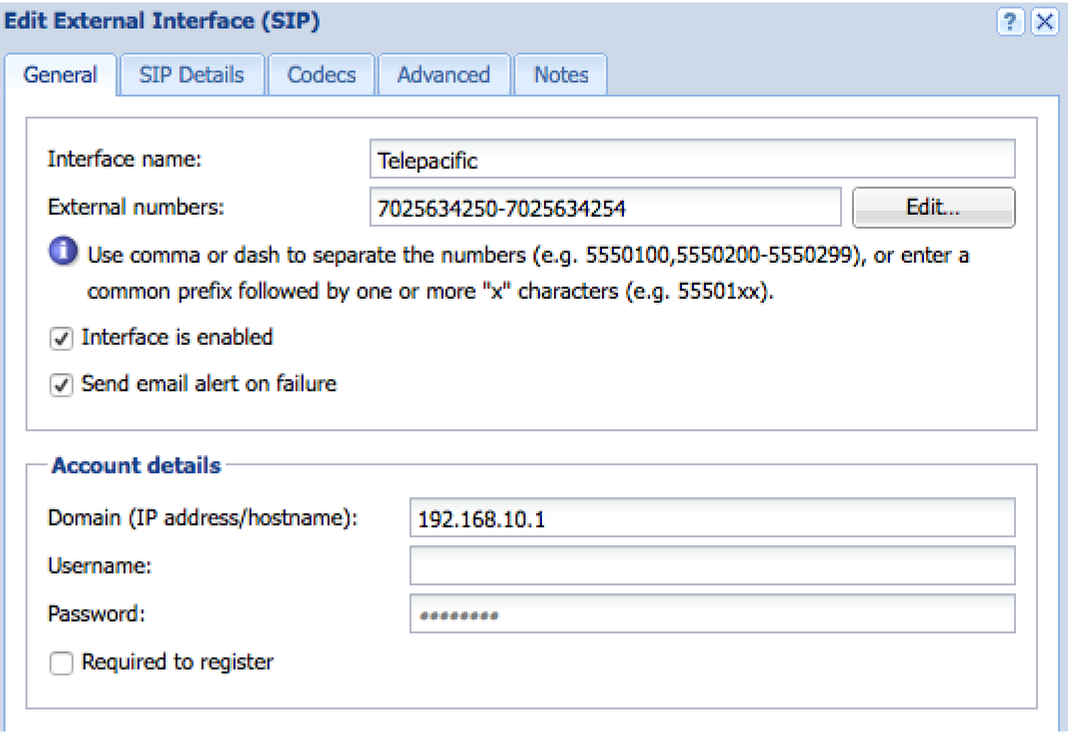
5. Choose the extension to receive incoming calls and leave the dial out prefix empty

6. Click **Next.**

7. Specify the IP address of the CPE Router (e.g., 192.168.12.1) and do not change the default port

8. Uncheck **Required to register** and leave the **Username** and **Password** fields empty

9. Click **Next** and **Finish**



2.7.14 Connecting Kerio Operator to Teliax

You can configure a SIP trunk with Teliax for dialing to the public telephone network. This topic describes the necessary configuration in Kerio Operator. For more information, refer to Connecting to VoIP service providers (page 25).

Prerequisites

Teliax requires information about the IP address of your Kerio Operator instance in order to ensure a secure connection.

After setting up an account with Teliax, you should be given the following information from the provider, which is required in this configuration:

- Server (IP or Domain such as test.ivy.teliax.com)
- Login ID
- Password
- DID/Telephone Number
- Channels (not needed for Operator configuration)

Configuration

1. Log in to the web administration interface of Kerio Operator.

2. Go to **Configuration > Call Routing**.

3. Click **Add a SIP Interface**.

4. Enter an interface name. For example, `Teliax`.

5. Choose **New provider** and enter your telephone number (you may need to add a `1` at the beginning of the telephone number). Use a comma to separate multiple phone numbers.

6. Click **Next**.

7. Choose the extension to receive incoming calls.

8. Enter a dial out prefix if necessary.

9. Click **Next**.

10. Specify the hostname (Server) provided by Teliax (`test.ivy.teliax.com`) and the default port `5060`.

11. Specify the **Username** (Login) and **Password** values as provided by Teliax.

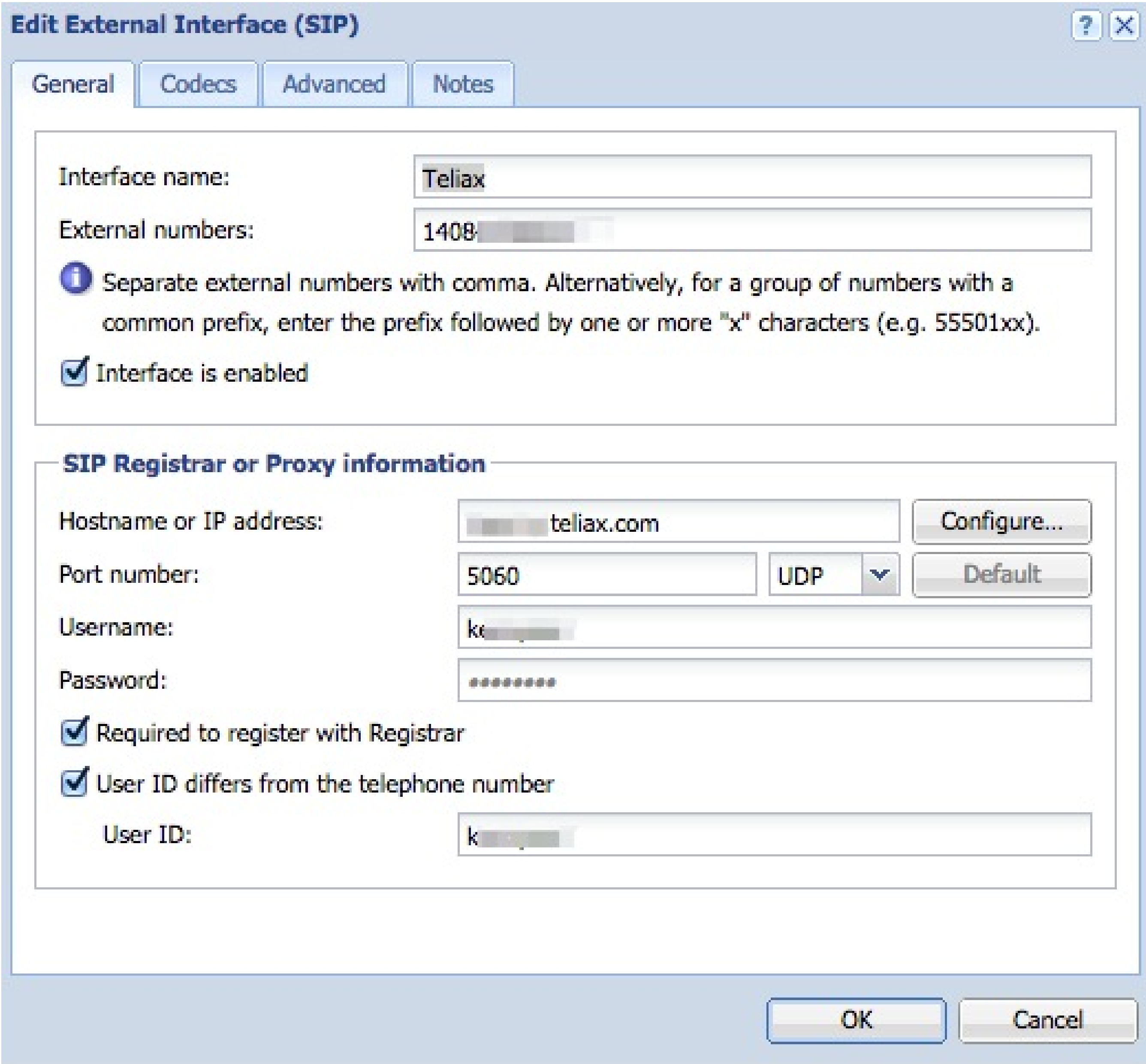
12. Enable **User ID differs from the telephone number** and enter your Teliax User ID (the same number that is in the Username field).

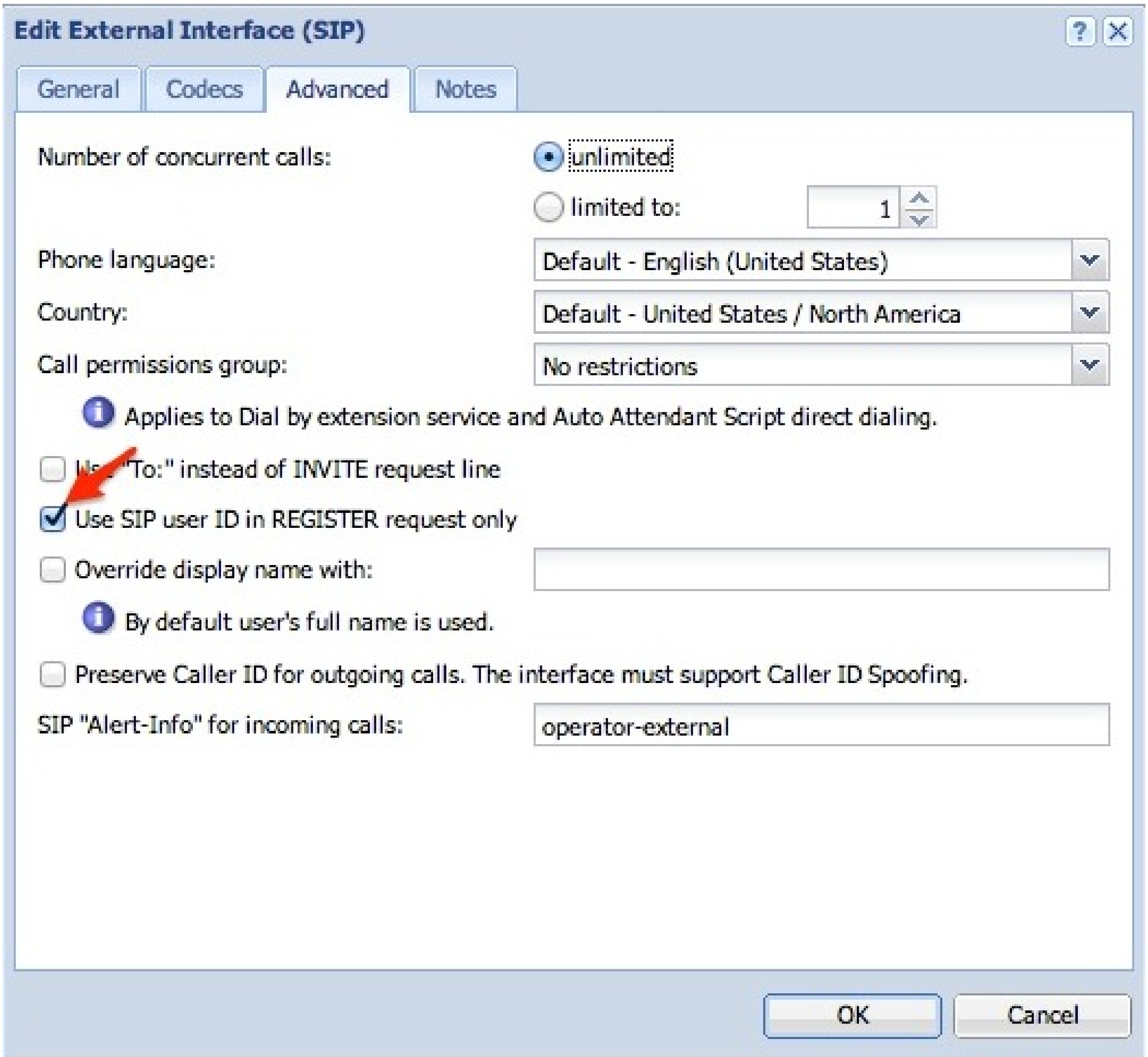
13. Enable the option **Register with registrar**.

14. Click **Finish**.

15. In order to transmit the correct CallerID information on outbound calls you must edit the interface that you've just created by doing the following:

- Select the interface you just created (e.g. Teliax) and click **Edit**.
- Click the **Advanced** tab.
- Check the box **Use SIP user ID in REGISTER request only**.
- Click **OK** to save.


Edit External Interface (SIP)
General
Codecs
Advanced
Notes
Interface name:
Teliax
External numbers:
1408
Separate external numbers with comma. Alternatively, for a group of numbers with a common prefix, enter the prefix followed by one or more "x" characters (e.g. 55501xx).
Interface is enabled
SIP Registrar or Proxy information
Hostname or IP address:
teliax.com
Configure...
Port number:
5060
UDP
Default
Username:
Password:
Required to register with Registrar
User ID differs from the telephone number
User ID:
OK
Cancel



2.7.15 Connecting Kerio Operator to Vitelity

> **NOTE**
> This information is designed for Kerio Operator 2.3.5 and older.

You can configure a SIP trunk with Vitelity for dialing to the public telephone network. This topic describes the necessary configuration in Kerio Operator. For more information refer to Connecting to VoIP service providers.

Prerequisites

After setting up an account with Vitelity, you should be given the following information from the provider:

- **Service**: Vitelity LLC VoIP
- **Username**: *<your_user_name>

» **Password**: *<your_password>

» **Balance**: $<account_balance>

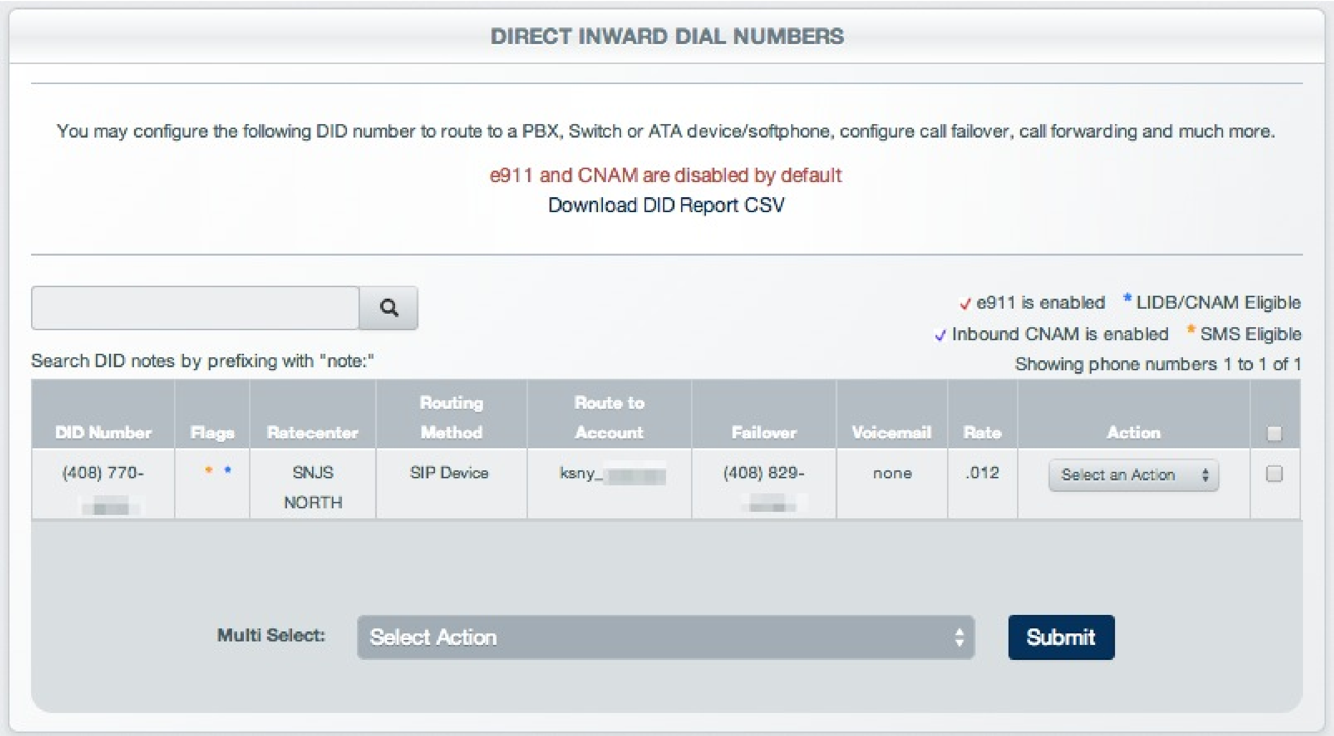
The username and password above are for managing your Vitelity account. They are not used within Kerio Operator for any reason.

You then need to set up the following on the Vitelity admin console:

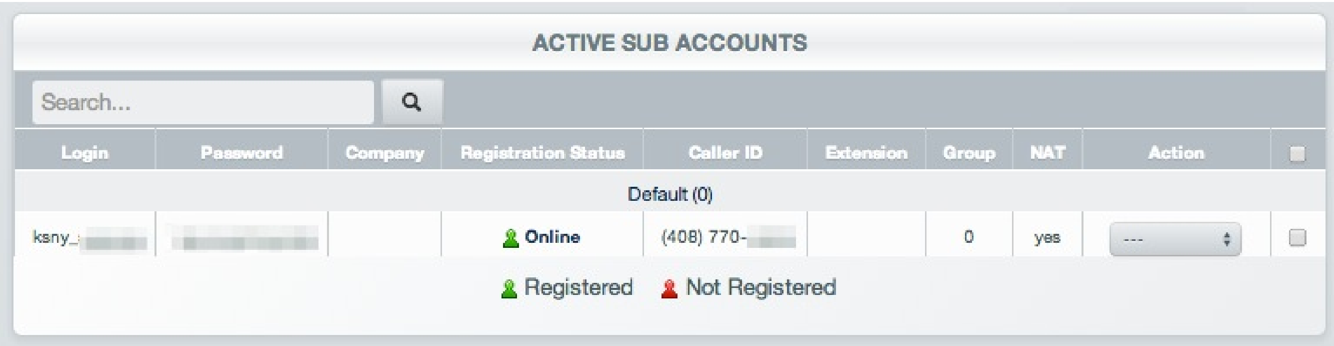
» A **Sub Account** (your SIP account).

» A DID Number.

The following screenshots serve as a guide:



Screenshot 3: Figure 1 – Vitelity DID Number Configuration



Screenshot 4: Figure 2 - Vitelity Sub Account (SIP) Configuration

After all the settings as described above, you need the following information for configuring Kerio Operator:

» The telephone number (or numbers) assigned to you.

» Login (sometimes referred to as Trunk #, User Name or User ID).

» Password

» Register Server (`inbound34.vitelity.net`).

» Outbound Server (`outbound.vitelity.net`).

» Additional information needed for configuration under **Support > Generic Sip Support** (for example, a different Register Server: `sip34.vitelity.net)`.

Configuration

1. Log in to the web administration interface of Kerio Operator.

2. Go to **Configuration Call Routing**.

3. Click **Add SIP Interface**.

4. Enter an interface name (e.g. "**Vitelity**").

5. Choose **New Provider** and enter your telephone number (you may need to add a `1` at the beginning of the telephone number). Use a comma to separate multiple phone numbers.

6. Click **Next**

7. Choose the extension to receive incoming calls

8. Enter a dial out prefix if necessary.

9. Click **Next**.

10. Specify the hostname provided by Vitelity under **Support > Generic Sip Support** (`sip34.vitelity.net`) and the default port 5060.

11. Specify the **Username** (login and password values as you configured on Vitelity's sub account settings (see Figure 2 above).

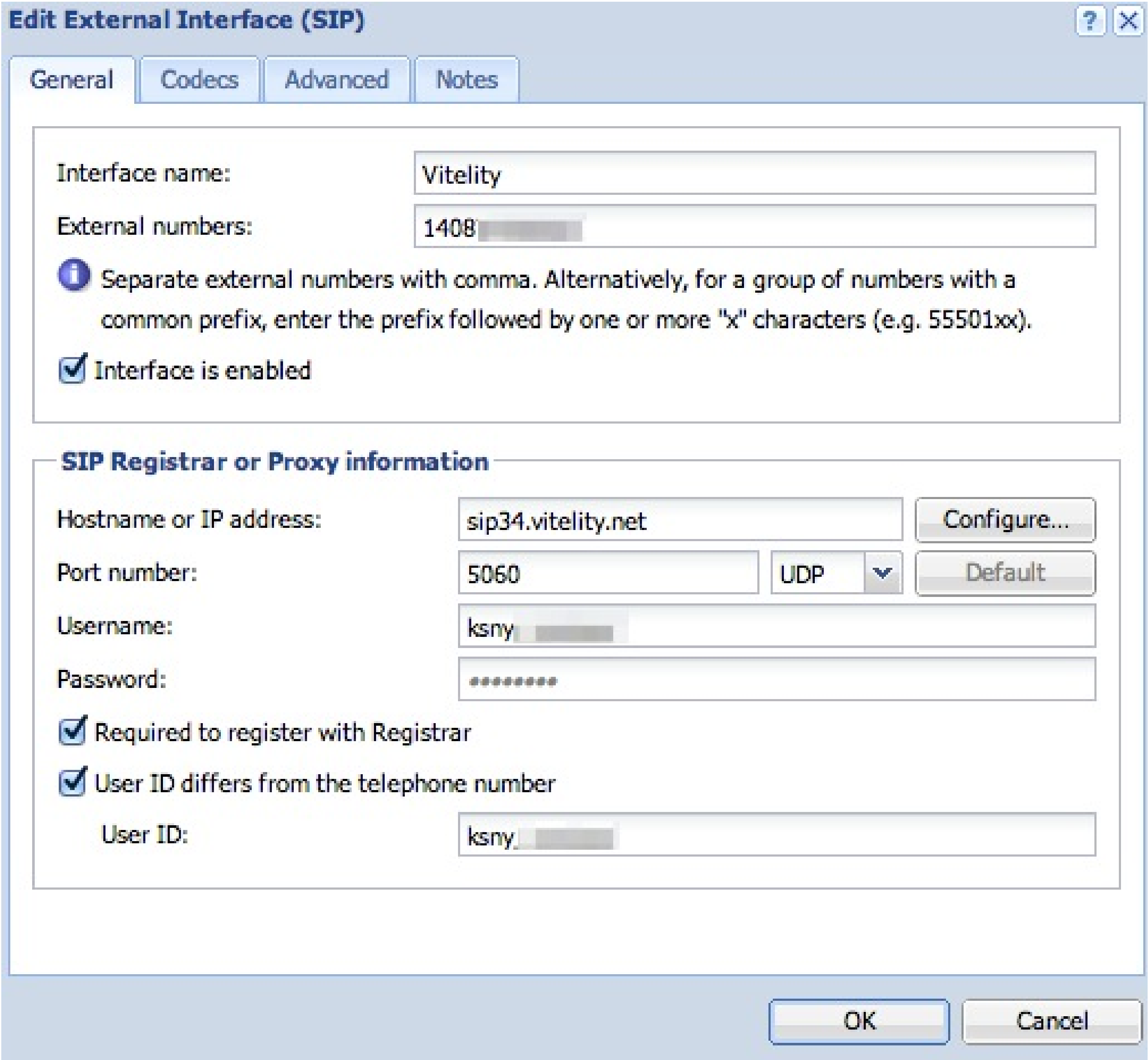
12. Enable **User ID differs from the telephone number** and enter your Vitelity login (the same string that is in the username field).

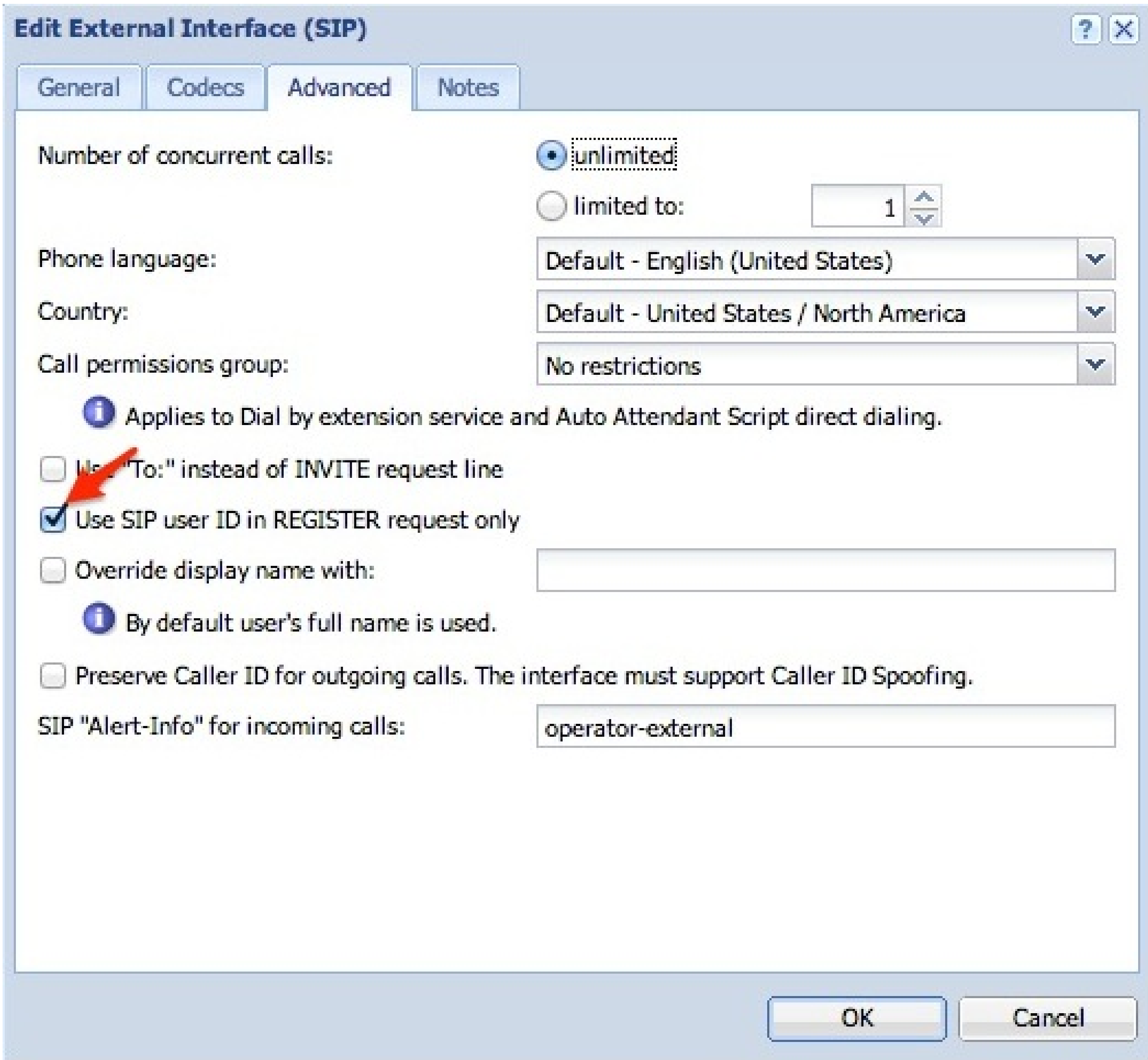
13. Enable the option to **Register with registrar**.

14. Click **Finish**.

15. In order to transmit the correct CallerID information on outbound calls, you must edit the interface that you created. Do the following:

- Select the interface you just created (e.g. Vitelity) and click **Edit**.
- Click the Advanced tab
- Check the box Use SIP user ID in REGISTER request only
- Click OK to save


Edit External Interface (SIP)
General
Codecs
Advanced
Notes
Interface name:
Vitelity
External numbers:
1408
Separate external numbers with comma. Alternatively, for a group of numbers with a common prefix, enter the prefix followed by one or more "x" characters (e.g. 55501xx).
Interface is enabled
SIP Registrar or Proxy information
Hostname or IP address:
sip34.vitelity.net
Configure...
Port number:
5060
UDP
Default
Username:
ksny
Password:
Required to register with Registrar
User ID differs from the telephone number
User ID:
ksny
OK
Cancel



2.7.16 How to configure Kerio Operator to connect to 802.cz

NOTE

This information is designed for Kerio Operator 2.4 and newer.

Prerequisites

You need the following information for this configuration:

- Your telephone number. You will have selected this number when you registered with 802.cz. You can also find the number if you log into your customer account at www.802.cz. The telephone number (in the national format, i.e. 9 digits) is also used as the authentication name.
- Your SIP password. You can configure this in your customer account at www.802.cz.

Configuration

1. Log in to the administration interface of your Kerio Operator and navigate to the **Call Routing** screen.

2. Click **Add a SIP Interface**.

3. In the first screen, enter a description for the interface name (for example, '802.cz'), choose the option **New provider** and enter your telephone number in the national format (9 digits). For example, if your telephone number is in the international format `+ 420 333 123 456`, you would enter `333123456`.

4. Click **Next**.

5. Choose the extension to receive incoming calls and enter the prefix that will be used to dial out (for example, 0. The usual dial-out prefix in the Czech Republic).

6. Click **Next**.

7. Enter `hlas.802.cz` as the **Domain (IP address/hostname)**. If you have a nomadic phone number, use `sip.802.cz` instead.

8. Enter your telephone number in the national format (9 digits, e.g. `333123456`) as the **Username** and enter your SIP password as the **Password**.

9. Ensure **Required to register** is checked and click **Next**.

10. Verify the information in the **Summary** section and click **Finish**.

Your **802.cz** connection is now configured. You can now test this, by placing some calls to your telephone number. This will verify that your SIP connection is working correctly.

2.7.17 How to configure Kerio Operator to connect to ActiveNetwork

NOTE
This information is designed for Kerio Operator 2.4 and newer.

Learn how to configure connect **Kerio Operator** to Active Network using either a SIP truck also known as `postpagato` or, using account `prepagato`.

Prerequisites

Before configuring **Kerio Operator**, you will require the following information:

- Telephone number, as provided by Active Network in the **Area Riservata** on their website.
- User-name of the SIP account.
- Password
- SIP Domain
- The SIP account type - `postpagato` or `prepagato`.
- If you have a firewall, make sure the SIP and RTP ports are open and properly routed to **Kerio Operator**.

Configuration

This process describes how to connect **Kerio Operator** to a SIP account with Active Network.

1. Log in to the administration interface of your **Kerio Operator** and navigate to the **Call Routing** screen.

2. Click **Add a SIP Interface**.

3. Enter a description for the interface name. For example, `activenetwork-prepagato`).

4. Choose the option **New provider** and enter your telephone number.

   - With the option `postpagato`, it is necessary have the number **5** in front of the phone number. For example, 507611878139.

   - With the option `prepagato`, the phone number doesn't have any prefix in front of the number. For example, 07611763399.

5. Click **Next**.

6. Choose the extension to receive incoming calls and enter the prefix that will be used to dial out.

7. Click **Next**.

   a. Enter the Domain (IP address/hostname).

      - For an account `prepagato` use **VOIP.EUTELIA.IT**.

      - For an account `postpagato` use **SIP.TWT.IT**.

   b. Enter your **Active Network User Name** as the username and the **SIP-Password** as the password.

   c. Ensure **Required to register** is checked.

8. Click **Next** to continue to the **Summary** screen.

9. Verify the information and click **Finish**.

2.7.18 How to configure Kerio Operator to connect to Bandwidth.com

Learn how to configure Kerio Operator for use with a SIP trunk from Bandwidth.com.

> **NOTE**
> This information is designed for Kerio Operator 2.4 and newer.

> **NOTE**
> Bandwidth.com requires that you run your VoIP PBX on a public IP address. Ensure that your Kerio Operator installation is secure.
> Use strong passwords and configure the built-in firewall. Only your own phones and Bandwidth.com's server should be allowed to communicate with your Kerio Operator PBX. With Kerio Operator 1.1, use the protection against SIP password guessing.

Prerequisites

Before starting this procedure, ensure you have:

» Your account is configured for E.164 dialing (the international format with + at the beginning). In this topic, we assume you use E.164 which is the default and also a slightly complex case.

» The telephone number assigned to you by Bandwidth.com. For example, `+1 234 555 0101`.

» The IP address of Badwidth.com's primary gateway. For example, `216.82.224.202`.

Configuration

1. Log in to the administration interface of your Kerio Operator and go to the **Call Routing** screen.

2. Click **Add a SIP Interface**.

3. Key in the interface name. For example, `bandwidth.com`.

4. Choose the option **New provider** and key in your telephone number in the international format with `+1` at the beginning. For example, `+12345550101`.

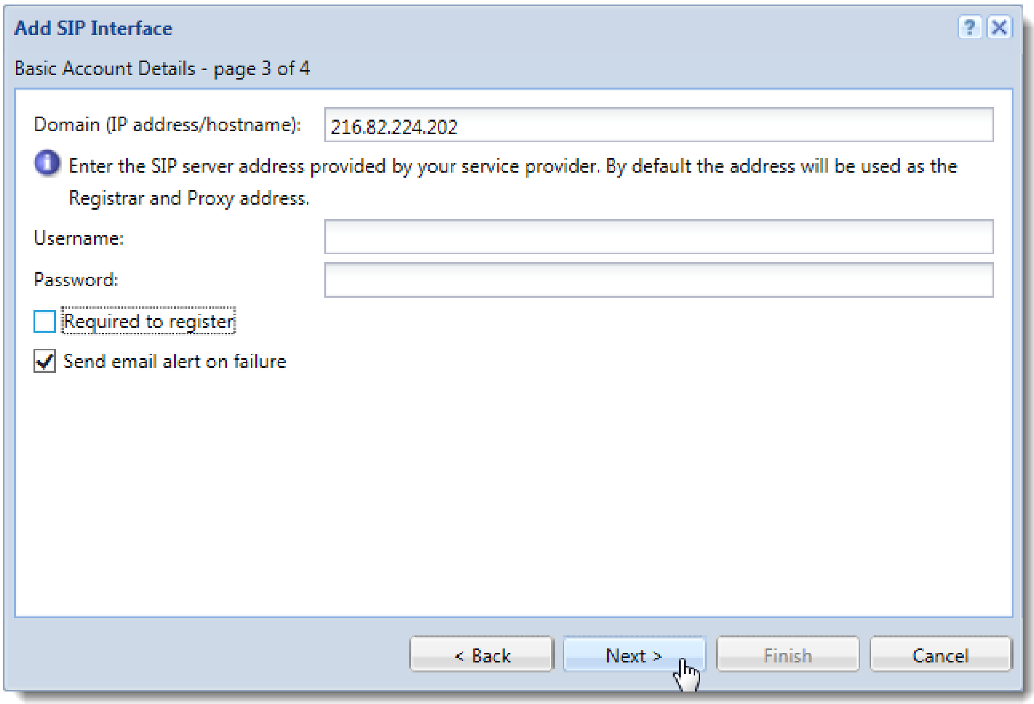
5. Click **Next**.

6. Choose the extension to receive incoming calls and key in the prefix that will be used to dial out. For example, `9`.

7. Click **Next**.

8. Key in the IP address (`216.82.224.202`) of Bandwidth's primary gateway in the **Domain (IP address/hostname)** field.

9. Uncheck **Require to register**, and leave the **username** and **password** fields empty. Bandwidth.com uses only IP-based authentication, so you must register your IP address with them.



10. Click **Next**.

11. Verify the information in the **Summary** section and click **Finish** to save the configuration of the interface.

You should now be able to receive incoming calls. But we need to modify the rules for outgoing calls so that outgoing calls work as well and dialed numbers are rewritten to the E.164 format.

We assume that if the PBX user dials a number with `91` at the beginning, they want to place a domestic call. If the number starts with `9011`, it's an international call.

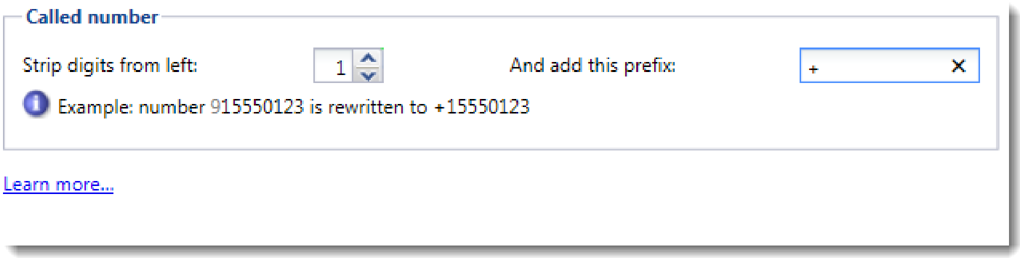
Dialing a + on a desktop phone is usually not easy, so we will add the + sign in the rewriting rules instead.

1. Double-click on the rule for outgoing calls to edit it. In our example, the rule is displayed as `9... bandwidth.com`.

2. In the edit dialog, change the prefix to 91 and go to the **Called number** section.

3. Set **Strip digits from left** to 1.

4. In the **And add this prefix** field, key in +. Now, if someone dials 914084964500, this number will be modified to +14084964500. This ensures we dial US-based numbers in the full E.164 format.

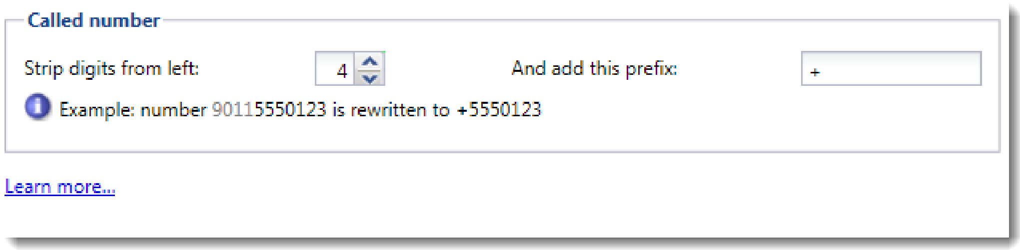


5. Click **OK** to save the change.

6. Click **Add...** on the **Call Routing** screen to add a second outgoing rule for prefix 9011.

7. Key in 9011 as the prefix and add the external interface for Bandwidth.com.

8. Under **Called Numbers**, set **Strip digits from left** to 4 and set **And add this prefix** to +. On setting these fields, the prefix 9011 will be replaced with + and we will thus obtain a correct international E.164 format.



9. Click **OK** to save the dialog.

Your Bandwidth.com trunk is now configured on Kerio Operator. Try running some test calls.

2.7.19 How to configure Kerio Operator to connect to Breezz (NL)

Learn how to configure **Kerio Operator** to connect to the Dutch provider Breezz.

> **NOTE**
> This information is designed for Kerio Operator 2.4 and newer.

Prerequisites

Before starting this procedure, ensure you have:

» The telephone number assigned to you by **Breezz**. The number will include the Netherlands' international country code (without + at the beginning). For example, 31718123456. In the configuration, the phone number is also used as an authentication name.

» The password for your SIP account.

Configuration

1. Log in to the administration interface of **Kerio Operator**.

2. Go to **Advanced Options**, tab **General** and set the **SIP User-Agent** string to **Kerio Operator**. Breezz will not let you register with the **Asterisk PBX** string.

3. Go to the **Call Routing** screen.

4. Click **Add a SIP Interface**.

5. In the first screen, enter the interface name (for example, `Breezz-NL`), choose the option **New provider** and enter your telephone number (`31718123456` as per our example).

6. Click **Next**.

7. Choose the extension to receive incoming calls and enter the prefix that will be used to dial out. For example, `9`.

8. Click **Next**.

9. Enter `sip.sipnl.net` to the Domain (IP address/hostname).

10. Enter your telephone number (`31718123456`) as the username.

11. Enter your SIP password.

12. Ensure **Require to register** is checked.

13. Click **Next** to go to the **Summary** screen.

14. Verify the information and click **Finish**.

15. Double-click the interface you have just created to edit the order of codecs.

16. Move **G.711 A-law** and **G.711 U-law** to the top. It is recommended to this because **Breezz** now supports **G.726** and the translation from **G.711** to **G.726** reduces the audio quality noticeably. If your phones support **G.726** you may leave the codec order as it is and use **G.726** for the whole call path (**phone < > Operator < > Breezz**).

You are now ready to place and receive calls.

2.7.20 How to configure Kerio Operator to connect to DevopSys

Learn how to configure Kerio Operator to connect to Devopsys using a SIP trunk.

> **NOTE**
> This information is designed for Kerio Operator 2.4 and newer.

Prerequisites

Before starting this procedure, ensure you have:

- Your telephone number.
- The Devopsys SIP Domain or IP address.
- Username and Password.
- If you have a firewall, make sure the SIP and RTP ports are open and properly routed to Kerio Operator.
- You must have communicated to Devopsys the IP address you are going to use to connect Operator to their services – If it is behind firewall the IP address of the firewall will be used.

Configuration

This process describes how to connect Kerio Operator to a SIP account with Devopsys.

1. Log in to the administration interface of your Kerio Operator and navigate to the **Call Routing** screen.

2. Click **Add a SIP Interface**.

3. In the first screen:

    a. Enter a description for the interface name. For example, `Devopsys`.

    b. Choose the option **New provider** and enter your telephone number including the international dialing code. For example `0033 5 87 030300`.

4. Click **Next**.

5. Choose the extension to receive incoming calls, and enter the prefix that will be used to dial out, if you wish to use one.

6. Click **Next**.

    a. Enter the SIP server in the **Domain (IP address/hostname)**.

    b. Enter your **Devopsys User Name** as the username and the **SIP-Password** as the password.

    c. Ensure **Required to register** is checked.

7. Click **Next** to continue to the **Summary** screen.

8. Verify the information and click **Finish**.

Your Devopsys connection is now configured.

2.7.21 How to configure Kerio Operator to connect to Exetel

Learn how to connect Kerio Operator to a SIP account with Exetel. Exetel can be reached at : http://www.exetel.com.au.

> **NOTE**
> This information is designed for Kerio Operator 2.4 and newer.

Prerequisites

Before starting this procedure, ensure you have:

- The telephone number assigned to you. The phone number will be used as an authentication name as well.
- The password for your SIP account.
- SIP proxy address (example Exetel SIP server, `sip1.exetel.com.au`.

Configuration

1. Log in to the administration interface of **Kerio Operator**.

2. Go to the **Call Routing** screen.

3. Click **Add a SIP Interface**.

4. In the first screen, enter the interface name. For example, `Exetel`.

5. Choose the option **New provider** and enter your telephone number.

6. Click **Next**.

7. Choose the extension to receive incoming calls (queue, script, conference or group) and enter the prefix that will be used to dial out. For example, 9.

8. Click **Next**.

9. Enter **Domain (IP address/hostname)**. For example, `sip1.exetel.com.au`.

10. Enter your telephone number as the **username** and your SIP password as **password**.

11. Ensure **Required to register** is checked.

12. Click **Next**.

13. Verify the information in the **Summary** section and click **Finish**.

You are now ready to place and receive calls.

2.7.22 How to configure Kerio Operator to connect to fayn.cz

Learn how to connect Kerio Operator to a SIP account with Fayn. We assume that you already have a Fayn SIP account and know your SIP credentials.

> **NOTE**
> This information is designed for Kerio Operator 2.4 and newer.

Prerequisites

If you do not know the credentials, log in to your account at www.fayn.cz, go to **Setup** and click **View** next to your SIP authentication name.

Before starting this procedure, ensure you have:

- Your phone number as provided by fayn.cz.
- Your password.
- SIP proxy address. Currently, `sip.fayn.cz`.

All of this information can be found at https://iz.fayn.cz/ after login, under the *Přehled MSN* menu and the *MSN* button.

Configuration

1. Log in to the Kerio Operator admin interface, go to the **Call Routing** section.

2. Click **Add a SIP Interface** button.

3. Name your new interface and enter your assigned phone number.

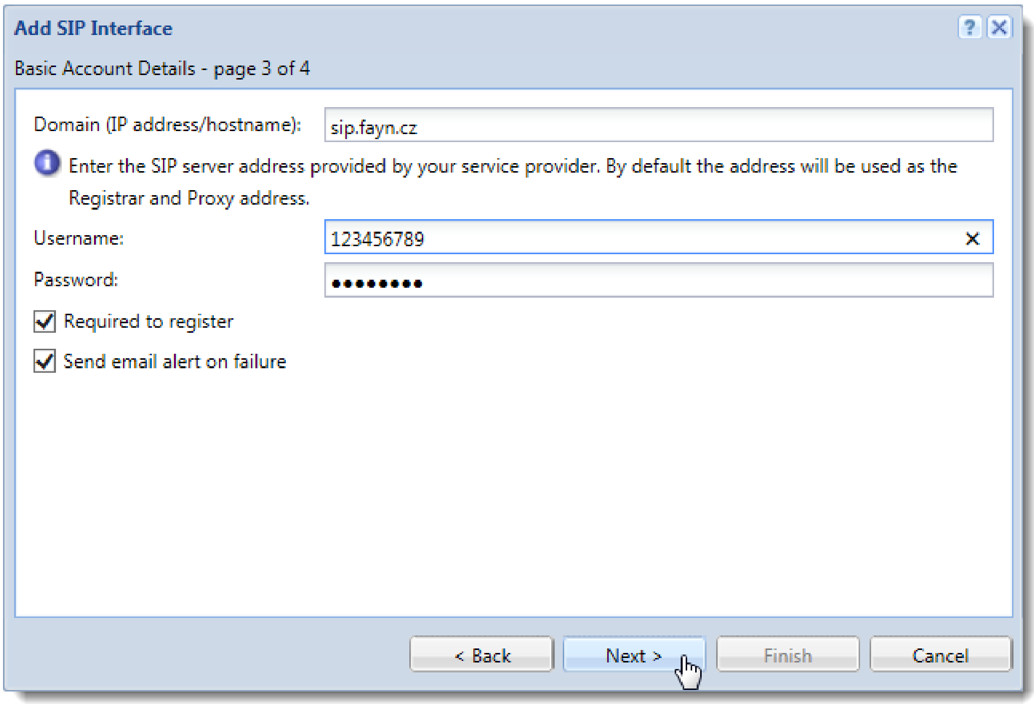
4. Click **Next**.

5. Select the desired internal extension (queue, script, conference or group) and optionally enter an outbound prefix.

6. Click **Next**.

7. Enter `sip.fayn.cz` into the **Domain (IP address/hostname)** field.

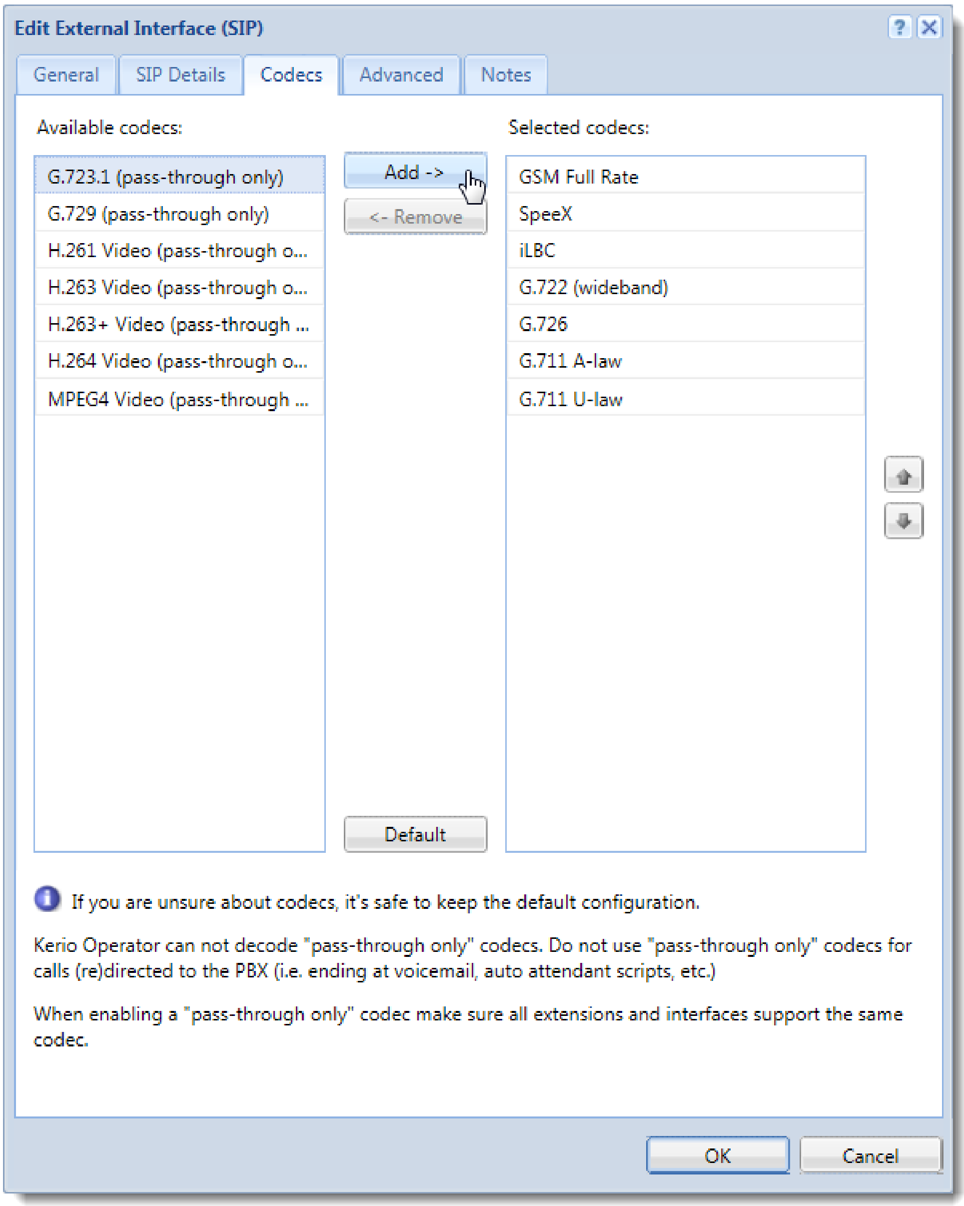
8. Enter **Username**(your assigned phone number) and your password.



9. Click **Next**.

10. Verify the information in the **Summary** section and click **Finish**.

11. Open the route configuration again, go to the **Codecs** tab and correct the supported codecs list to match those supported by Fayn (see your Fayn account for exact information: http://www.fayn.cz/pece-a-podpora/navody-a-nastaveni/).



12. Press **OK**

2.7.23 How to configure Kerio Operator to connect to ha-vel.cz

Learn how to connect Kerio Operator to a SIP account with ha-vel.cz.

> **NOTE**
> This information is designed for Kerio Operator 2.4 and newer.

Prerequisites

We assume that you already have a Ha-vel SIP account and know your SIP credentials. (If you do not know the credentials, login to your account at https://ha-loo.ha-vel.eu/cz/index.php).

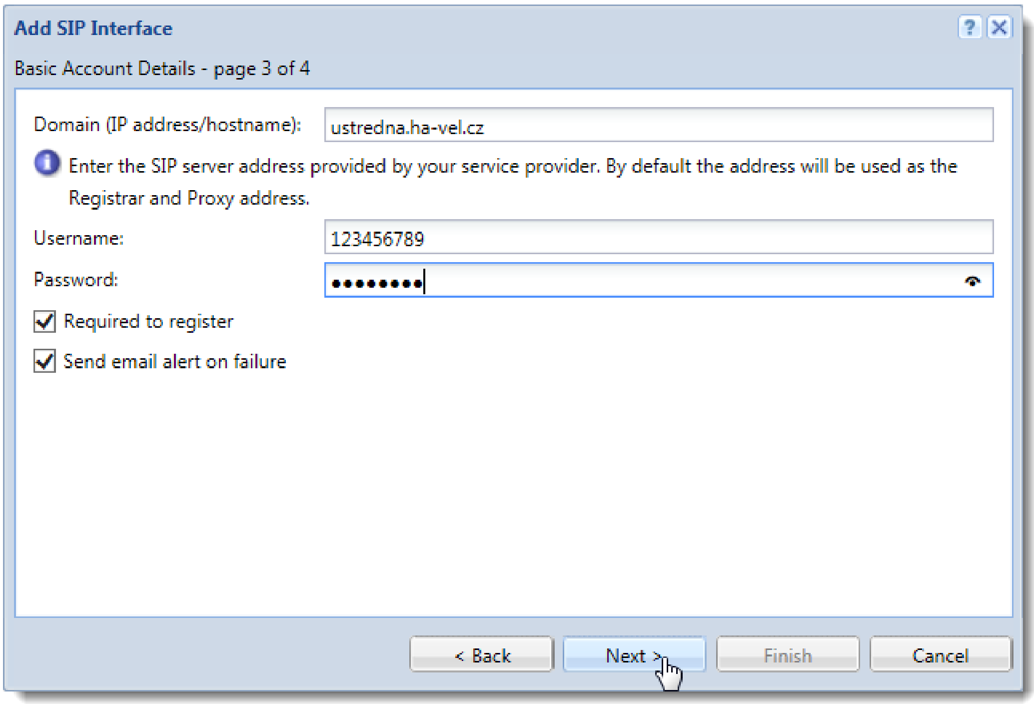
Before starting this procedure, ensure you have:

- Your phone number registered with ha-vel.cz. We assume it as `123456789` in our example below.
- Your password.
- SIP proxy address (currently `ustredna.ha-vel.cz`).

These details can be found by logging into https://ha-loo.ha-vel.eu/cz/index.php, under *Information / Informace* and *SIP/IAX Settings / Nastaveni SIP/IAX* sections .

Configuration

1. Log in to the Kerio Operator admin interface.
2. Go to the **Call Routing** section and click **Add a SIP Interface**.
3. Name your new interface and enter your assigned phone number.
4. Click **Next**.
5. Select the desired internal extension (queue, script, conference or group) and optionally specify the outbound prefix.
6. Click **Next**.
7. Enter `ustredna.ha-vel.cz` into the **Domain (IP address/hostname)** field.
8. Key in the assigned phone number into **Username** field and password into **Password** field.
9. Click **Next**.



10. Open route configuration again, go to the **Codecs** tab and remove unsupported codecs (SpeeX, G.722 and G.726 codecs).
11. Click **OK**

2.7.24 How to configure Kerio Operator to connect to isphone

> **NOTE**
> This information is designed for Kerio Operator 2.4 and newer.

Prerequisites

Before starting this procedure, ensure you have:

- The telephone number assigned to you in E164 format. The phone number will be used as an authentication name as well.
- The password for your SIP account.
- SIP proxy address. For example, `sip2.isphone.com.au`.

Configuration

1. Log in to the administration interface of **Kerio Operator**.
2. Go to the **Call Routing** screen.
3. Click **Add a SIP Interface**.
4. Key in the interface name (for example, `isphone`).
5. Choose the option **New provider** and enter your telephone number.
6. Click **Next**.
7. Choose the extension to receive incoming calls (queue, script, conference or group) and enter the prefix that will be used to dial out. For example, 9.
8. Click **Next**.
9. Set the **Domain (IP address/hostname)**, For example, `sip2.isphone.com.au`.
10. Enter your telephone number as the **username** and your SIP password as **password**.
11. Ensure **Required to register** is checked.
12. Click **Next**.
13. Verify the information in the **Summary** section and click **Finish**.

You are now ready to place and receive calls.

2.7.25 How to configure Kerio Operator to connect to Megapath

> **NOTE**
> This information is designed for Kerio Operator 2.3.5 and older. For more information about creating and configuring a SIP interface in later versions, see Connecting to VoIP service providers.

Learn how to configure Kerio Operator with a Megapath SIP trunk.

Prerequisites

Before starting this procedure, ensure you have:

- Your telephone number(s).
- The User-Name of the SIP account.
- The password for the SIP account.
- The host or IP of the SIP registration server (usually an EdgeMarc device).

Configuration

1. Log in to the administration interface of your Kerio Operator and navigate to the **Call Routing** screen.
2. Click **Add a SIP Interface**.
3. Enter a description for the interface name. For example, `megapath`.
4. Choose the option **New provider** and enter your telephone number(s). For example, `4085555555`.
5. Click **Next**.
6. Choose the extension to receive incoming calls and an optional outgoing prefix.
7. Click **Next**.
8. Enter the SIP registrar/proxy hostname (e.g. `192.168.1.2`).
9. Keep the default port number, `5060`.
10. Enter the username and password values as provided to you by Megapath.
11. Ensure **Must register with the Registrar** is checked.
12. Enable **User ID differs from the telephone number**, and specify the same value as your username
13. Click **Finish**.
14. **Edit** your newly created SIP interface and from the **Advanced** menu enable **Use SIP user ID in REGISTER request only** option.

2.7.26 How to configure Kerio Operator to connect to MultiVoice

Learn how to configure Kerio Operator to connect to MultiVoice using a SIP trunk.

> **NOTE**
> This information is designed for Kerio Operator 2.4 and newer.

Prerequisites

Before starting this procedure, ensure you have:

- Your telephone number.
- The User name and password of the SIP account.
- The SIP Domain.
- If you have a firewall, make sure the SIP and RTP ports are open and properly routed to Kerio Operator.

Configuration

1. Log in to the administration interface of your Kerio Operator and navigate to the **Call Routing** screen.
2. Click **Add a SIP Interface**.

3. Enter a description for the interface name. For example, `MultiVoice`.

4. Choose the option **New provider** and enter your telephone number including the international dialing code. For example, `0039 0171 699757`.

5. Click **Next**.

6. Choose the extension to receive incoming calls, and optionally enter the prefix that will be used to dial out.

7. Click **Next**.

8. Enter the SIP Domain in **Domain (IP address/hostname**. For example, `multivoice.multiwire.net`.

9. Enter your `MultiVoice User Name` as the **username** and the `SIP-Password` as the **password**.

10. Ensure **Required to register** is checked.

11. Click **Next**.

12. Verify the information in the **Summary** section and click **Finish**.

Your MultiVoice connection is now configured.

2.7.27 How to configure Kerio Operator to connect to netphone.cz

Learn how to connect Kerio Operator to a SIP account with Netphone CZ.

> **NOTE**
> This information is designed for Kerio Operator 2.4 and newer.

We assume that you already have a Netphone SIP account and know your SIP credentials. If you do not know the credentials to login to your account on www.netphone.cz, go to Setup and click View next to your SIP authentication name.

Prerequisites

Before starting this procedure, ensure you have:

- Your phone number, as provided by netphone.cz. For example, `123456789`.
- Your password.
- SIP proxy address. Currently, `sip1.netphone.cz`.

These details can be found after logging in to https://admin.netphone.cz/prihlaseni, under *Přehled tel. cisel*section and *MSN* button.

Configuration

1. Log in to the Kerio Operator admin interface, go to the **Call Routing** section and click **Add a SIP Interface**.

2. Name your new interface and enter your assigned phone number on the 1st screen.

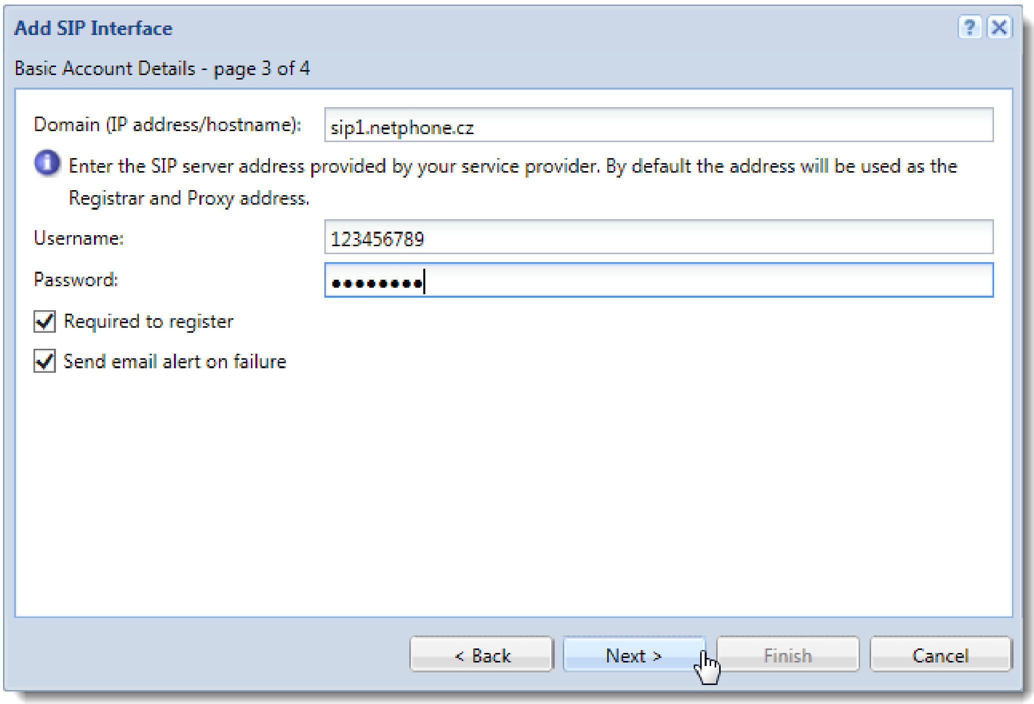
3. Click **Next**.

4. Select the desired internal extension (queue, script, conference or group) and optionally choose an outbound prefix.

5. Click **Next**.

6. Enter `sip1.netphone.cz` in the **Domain (IP address/hostname)** field.

7. Key in the **Username** (assigned phone number) and your **Password**.



8. Click **Next**.

9. Verify the information in the **Summary** section and click **Finish**.

10. Open route configuration and go to the **Codecs** tab and correct supported codecs list to match those supported by Netphone. See your netphone.cz account, under the *Prehled tel. cisel* section, click **Upravit** button and enable **Expert mod** to get the list.

11. Click **OK**

2.7.28 How to configure Kerio Operator to connect to OrbTalk

> **NOTE**
> This information is designed for Kerio Operator 2.3.5 and older. For more information about creating and configuring a SIP interface in later versions, see Connecting to VoIP service providers.

Learn how to configure connect Kerio Operator to OrbTalk using a SIP truck.

Prerequisites

Before starting this procedure, ensure you have:

» Your telephone number.

» The user-name and password of the SIP account.

» The SIP Domain

» If you have a firewall, make sure the SIP (`UDP 5060 – 5090`) and RTP (`UDP 6000 – 65535`) ports are open and properly routed to Kerio Operator. Kerio Operator requires that ports `UDP 10000 – 19999` internally are open.

Configuration

1. Log in to the administration interface of your Kerio Operator and navigate to the **Call Routing** screen.

2. Click **Add a SIP Interface**.

3. Enter a description for the interface name. For example, `OrbTalk`.

4. Choose the option **New provider** and enter your telephone number. For example, `0550935405`.

5. Click **Next**.

6. Choose the extension to receive incoming calls and enter the prefix that will be used to dial out.

7. Click **Next**.

8. Enter the SIP **registrar/proxy hostname**.

9. Enter `5060` in the **Port Number**.

10. Enter your OrbTalk User Name as the authentication name and the SIP-Password as the password. Typically the username will follow the format of `IPT-<country code and telephone number>`. For example, `IPT-441223202130`.

11. Ensure **Must register with the Registrar** is checked.

12. Enable **User ID differs from the telephone number**, this will also be your User Name.

13. Click **Finish**.

Your OrbTalk connection is now configured

2.7.29 How to configure Kerio Operator to connect to plusTEL in Denmark

Learn how to connect **Kerio Operator** to a SIP account with plusTEL.

> **NOTE**
> This information is designed for Kerio Operator 2.4 and newer.

Prerequisites

We assume that you already have a plusTEL SIP account and know your SIP credentials. (If you do not know the credentials, contact plusTEL.dk to receive this information.)

Before starting this procedure, ensure you have:

» Your SIP authentication name.

» Your SIP password.

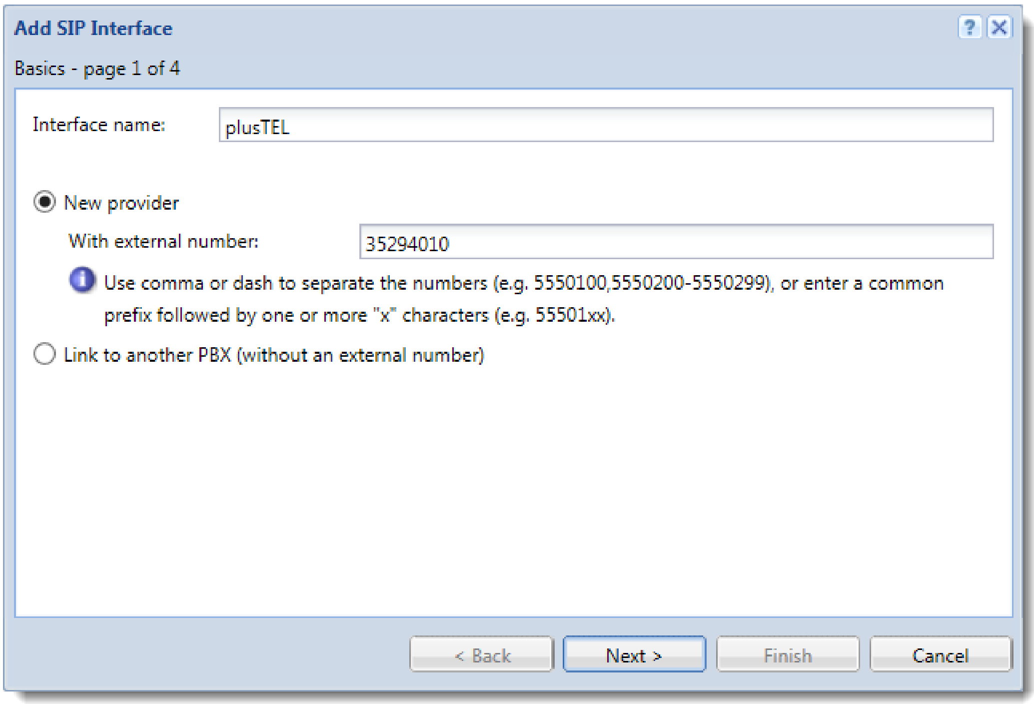
» The SIP registrar/proxy hostname.

Configuration

1. Log in to the administration interface of your **Kerio Operator** and go to the **Call Routing** screen. At least one extension is needed before creating the SIP Interface.

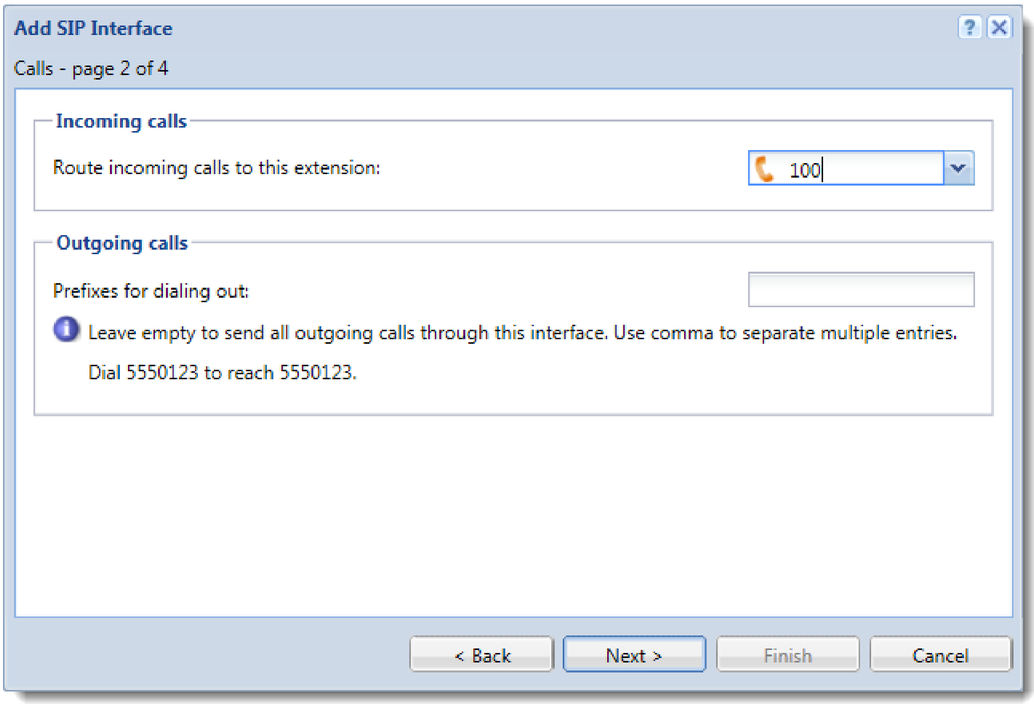
2. Click **Add a SIP Interface**.

3. Enter an interface name, choose the option **New provider** and enter your telephone number.



4. Click **Next**.

5. Choose the extension to receive incoming calls on and if you like, enter the prefix that will be used to dial out.

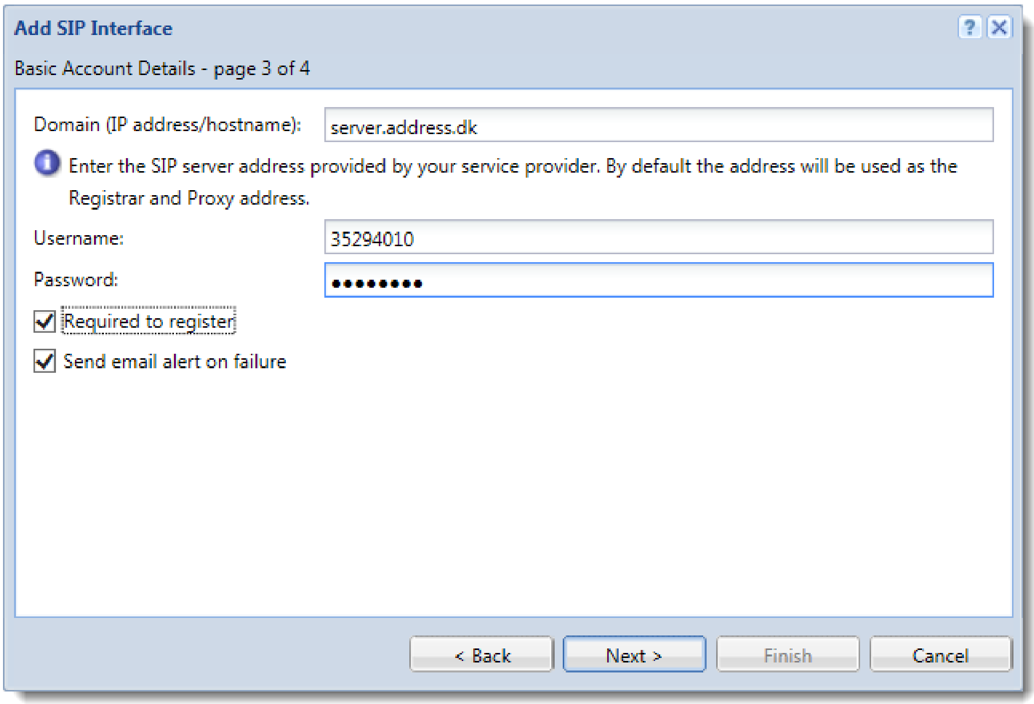


6. Click **Next**.

7. Enter the **Domain (IP address/hostname)**.

8. Enter your SIP **username** and **password**.

9. Ensure **Required to register** is checked.



10. Click **Next**.

11. Verify the information in the **Summary** section and click **Finish**.

Your connection to plusTEL is now configured.

plusTEL description

plusTEL is a registered VoIP Service Provider in Denmark.

For further information contact:

SIP-Trunk

plusTEL ApS

TEL: +45 35294010

info@plustel.dk

Kerio Distribution

MikroGraf as

TEL: +45 70222101

info@mikrograf.dk

eStation ApS

TEL: +45 35294000

info@estation.dk

2.7.30 How to configure Kerio Operator to connect to sipgate.com

Learn how to connect Kerio Operator to a SIP account with sipgate.com.

NOTE
This information is designed for Kerio Operator 2.3.5 and older. For more information about creating and configuring a SIP interface in later versions, see Connecting to VoIP service providers.

Prerequisites

Before starting this procedure, ensure you have:

» Your telephone number, as provided by Sipgate. To find the number, log in to Sipgate administration interface at www.sipgate.com and go to **Settings**.

» The SIP-ID of your SIP account. You can locate this in the Sipgate administration interface, click **Settings > SIP Credential**, the SIP-ID will then be displayed. For example, `1234567a0`.

» The SIP-ID (not the telephone number) is the most important identifier you require in order to connect over SIP. The SIP-Password for your SIP account. This password is usually of the form `1ABCD2`.

Configuration

1. Log in to the administration interface of your Kerio Operator and navigate to the **Call Routing** screen.

2. Click **Add a SIP Interface**.

3. Enter a description for the interface name. For example, `sipgate.com`.

4. Choose the option **New provider** and enter your telephone number. With sipgate, it is not significant whether you include `1` at the beginning of the number or not.

5. Click **Next**.

6. Choose the extension to receive incoming calls to and enter the prefix that will be used to dial out. For example, `9`.

7. Click **Next**.

8. Enter `sipgate.com` as the SIP registrar/proxy hostname.

9. Enter your sipgate SIP-ID as the authentication name and the SIP-Password as the password.

10. Ensure **Must register with the Registrar or Proxy** is checked.

11. Check **User ID differs from the telephone number** and enter your SIP-ID in the User ID field. This is the part where Sipgate differs from most SIP providers who prefer to use the telephone number as the SIP user ID.

12. Click **Finish**. Your sipgate connection is now configured.

You can now test the configuration by placing some calls to your telephone number. This will verify that your SIP connection is working correctly.

2.7.31 How to configure Kerio Operator to connect to Telephonic Canada

NOTE
This topic is for versions Kerio Operator 2.3.5 and older. For more information about creating and configuring a SIP interface in later versions, see Connecting to VoIP service providers.

Learn how to configure **Kerio Operator** with a SIP Trunk to Telephonic Canada

NOTE
Watch the Configuring Kerio Operator with Telephonic Canada video.

Prerequisites

Before starting this procedure, ensure you have:

- The telephone number or numbers assigned to you by Telephonic. Each number will include the Canadian' international country code without + at the beginning. For example, `17785555555`.
- If you have a firewall, make sure the SIP and RTP ports are open and properly routed to Operator.
- A static IP, as the typical configuration will use IP based registration without requiring authentication.
- You have provided Telephonic with your static IP address.

Configuration

1. Log in to the administration interface of **Kerio Operator**.
2. Go to the **Call Routing** screen.
3. Click **Add a SIP Interface**.
4. Enter the interface name. For example, `Telephonic`.
5. Choose the option **New provider** and enter your telephone number. For example, `17785555555`. In case of multiple numbers, use comma separation as noted in the dialog.
6. Click **Next**.
7. Choose the extension to receive incoming calls and enter the prefix that will be used to dial out. For example, 9.
8. Click **Next**.
9. In the SIP Registrar dialog, specify the hostname provided by Telephonic (`kerio.telephonic.ca`) and the default port `5060`. Leave the Username and password fields empty, and disable both checkboxes for registration and user ID.

NOTE
Telephonic Canada provides a free testing account (with DID) and $5.00 credit for customers evaluating Kerio Operator.

2.7.32 How to configure Kerio Operator to connect to Voicepulse.com

NOTE
This information is designed for Kerio Operator 2.4 and newer.

Learn how to connect Kerio Operator to a SIP account with Voicepulse Inc. We assume that you already have a Voicepulse SIP account and know your SIP credentials. (If you do not know the credentials, log in to your account at www.voicepulse.com, go to **Setup** and click **View** next to your SIP authentication name.

Prerequisites

Before starting this procedure, ensure you have:

» Your telephone number assigned to you by Voicepulse. For example `+1 555 123 4567`.

» Your SIP authentication name – note this is not the user name you use to log into Voicepulse web interface. The SIP authentication name is typically something like `VaSBFLM12a`".

» Your SIP password. Again, this is not the password you use to login to Voicepulse web interface. It is the second part of your SIP credentials. The password is typically something like `u2TYRUa878`.

Configuration

1. Log in to the administration interface of your Kerio Operator and go to the **Call Routing** screen.

2. Click **Add a SIP Interface**.

3. Enter a interface name, choose the option **New provider** and enter your telephone number with `1` at the beginning.

4. Click **Next**.

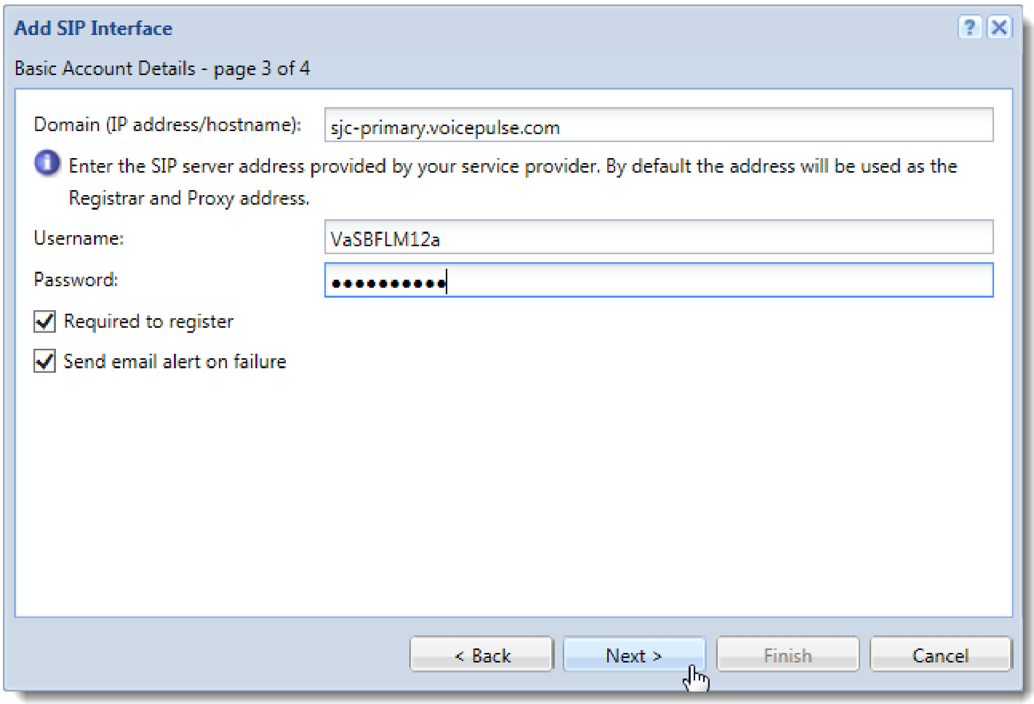
5. Choose the extension to receive incoming calls on and enter the prefix that will be used to dial out. For example, 9.

6. Click **Next**.

7. Enter `jfk-primary.voicepulse.com` as the **Domain (IP address/hostname)**. Use `sjc-primary.voicepulse.com` if you are closer to San Jose, CA than to New York.

8. Enter your SIP authentication name and password.

9. Ensure **Required to register** is checked.



10. Click **Next**.

11. Verify the information in the **Summary** section and click **Finish**.

Your connection to Voicepulse is now configured.

2.7.33 How to configure Kerio Operator to connect to VOIP-Unlimited

> **NOTE**
> This information is designed for Kerio Operator 2.4 and newer.

Learn how to connect Kerio Operator to a SIP account with VOIP Unlimited. We assume that you already have a VOIP unlimited SIP account and know your SIP credentials.

Prerequisites

Before starting this procedure, ensure you have:

- Your phone number and username, as provided by VOIP Unlimited.
- Your password.
- SIP proxy address. Currently, `sip.voip-unlimited.net`.

All of this information is emailed to you from VOIP-Unlimited.

Configuration

1. Log in to the Kerio Operator admin interface, go to the **Call Routing** section and click **Add a SIP Interface**.

2. Name your new interface and enter your assigned phone number.

3. Click **Next**.

4. Select the desired internal extension (queue, script, conference or group) and optionally enter an outbound prefix.

5. Click **Next**.

6. Enter `sip.voip-unlimited.net` in the **Domain (IP address/hostname)** field, enter your username (not your phone number) into the **Username** field and your password into the **Password** field.

7. Click **Next**.

8. Verify the information in the **Summary** section and click **Finish**.

9. Open the route configuration and go to the**Codecs** tab and correct the supported codecs list to match those supported by VOIP-Unlimited. The supported codecs are G729, G711a and G711u.

10. Click **OK**

2.7.34 How to configure Kerio Operator to connect to VoipVoice

> **NOTE**
> This information is designed for Kerio Operator 2.4 and newer.

Learn how to configure connect **Kerio Operator** to VoipVoice using a SIP truck.

Prerequisites

Before starting this procedure, ensure you have:

- Your telephone number.
- The user name and password of the SIP account.

- The SIP Domain.
- If you have a firewall, make sure the SIP and RTP ports are open and properly routed to **Kerio Operator**.

Configuration

1. Log in to the administration interface of your Kerio Operator and navigate to the Call Routing screen.
2. Click **Add a SIP Interface**.
3. Enter a description for the interface name. For example, `VoipVoice`.
4. Choose the option **New provider** and enter your telephone number. For example, `0550935405`.
5. Click **Next**.
6. Choose the extension to receive incoming calls and enter the prefix that will be used to dial out.
7. Click **Next**.
8. Enter the SIP domain in the **Domain (IP address/hostname)** field.
9. Enter your `VoipVoice User Name` as the **username** and the `SIP-Password` as the **password**.
10. Ensure **Required to register** is checked.
11. Click **Next**.
12. Verify the information in the **Summary** section and click **Finish**.

Your VoipVoice connection is now configured.

2.7.35 How to configure Kerio Operator to connect to Xphone.cz

> **NOTE**
> This information is designed for Kerio Operator 2.4 and newer.

Learn how to connect Kerio Operator to a SIP account with Xphone CZ. We assume that you already have a Xphone SIP account and know your SIP credentials. (If you do not know the credentials, login to your account at www.xphone.cz, go to **Setup** and click **View** next to your SIP authentication name.

Prerequisites

Before starting this procedure, ensure you have:

- Your phone number, as provided by xphone.cz. For exzample, `123456789`.
- Your username and password.
- SIP proxy address. Currently, `sip2.xphone.cz`.

All of this information can be found on logging in to http://www.xphone.cz, under *Nastaveni / Settings* and then *Prihlasovaci udaje / login* credentials.

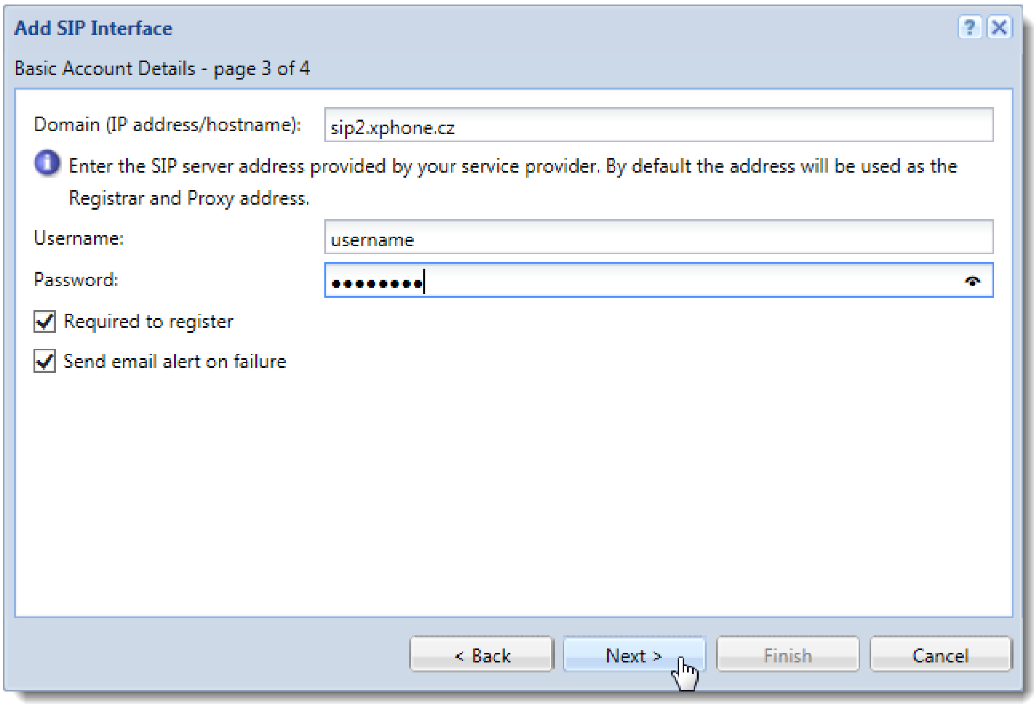
Configuration

1. Log in to the Kerio Operator admin interface, go to **Call Routing** section and click **Add a SIP Interface**.
2. Name your new interface and enter your assigned phone number.
3. Click **Next**.

4. Select the desired internal extension (queue, script, conference or group) and optionally enter an outbound prefix.

5. Click **Next**.

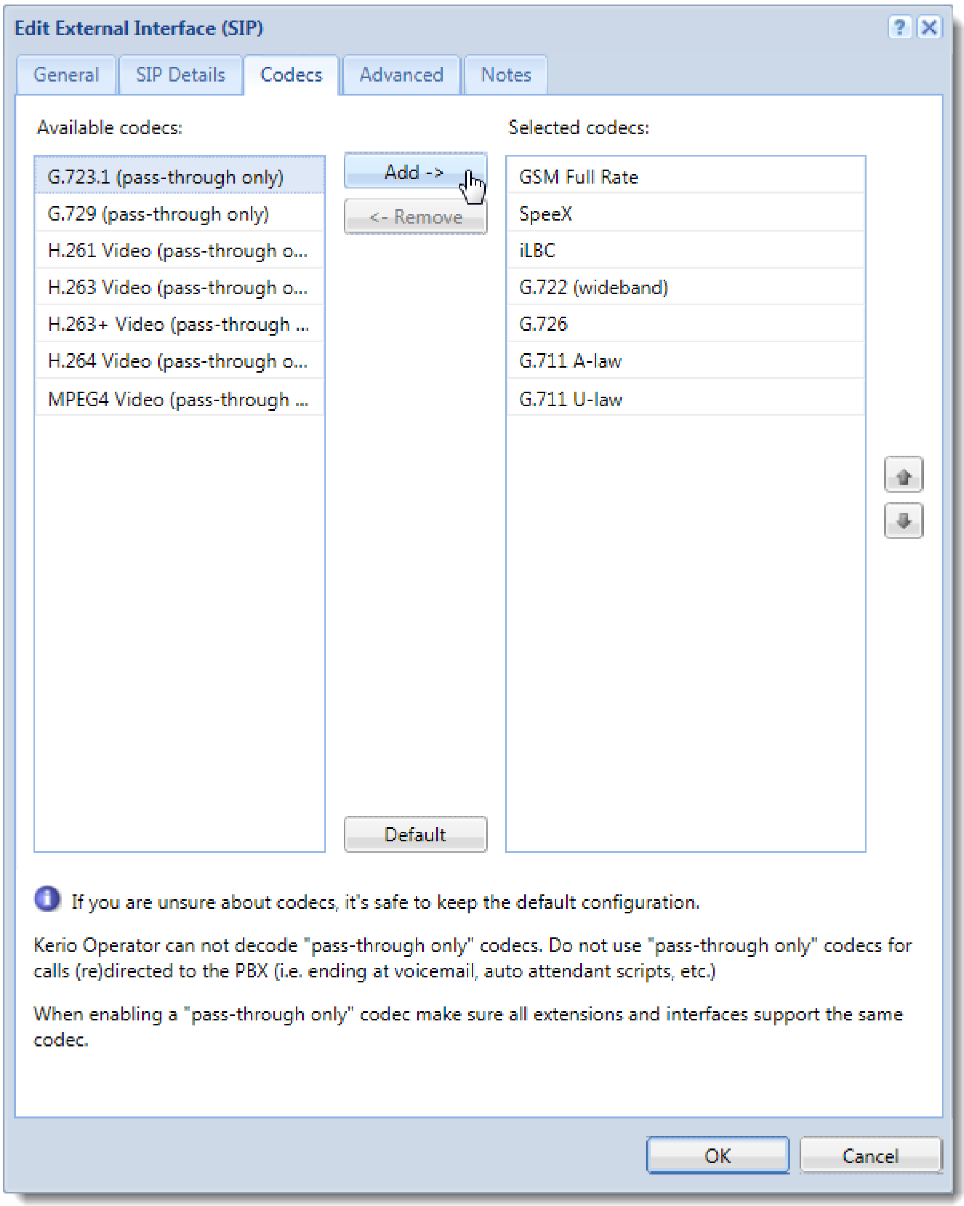
6. Enter `sip2.xphone.cz` in **Domain (IP address/hostname)**, enter your username (not your phone number) in **Username** and your password in **Password**.



7. Click **Next**.

8. Verify the information in the **Summary** section and click **Finish**.

9. Open the route configuration and go to the **Codecs** tab and correct the supported codecs list to match those supported by Xphone. See your xphone.cz account for exact information - https://www.xphone.cz/zone_user_settings_codecs.php).



10. Press **OK**

2.7.36 How to connect Kerio Operator to Skype Connect

> **NOTE**
> This functionality was last redesigned for Kerio Operator 2.4 and newer.

Prerequisites

Before starting this procedure, ensure you have:

» The SIP User ID for your Skype Connect profile. To find this ID, log to your Skype Manager account, view the details of your Skype Connect profile and navigate to Authentication Details. The SIP User is usually a 14-digit number that starts with `9905`.

» The password generated by Skype for your Skype Connect profile. You will find it in the same Skype Manager page as the SIP User ID.

» The phone number you have associated with Skype Connect profile. Skype lets you choose a number in about 20 countries. For example, a phone number in USA would look like `+1 408 555 0123`.

Configuration

1. Log in to the administration interface of your Kerio Operator and go to the **Call Routing** screen.

2. Click **Add a SIP Interface**.

3. Enter an interface name, choose the **New provider** option and enter your telephone number. For example, `14085550123`.

4. Click **Next**.

5. Choose the extension to receive incoming calls and enter a prefix that will be used for external calls. For example, `9`.

6. Click **Next**.

7. In the **Domain (IP address/hostname)** field, enter `sip.skype.com`.

8. Enter your SIP User ID in the **Username** field and enter the **Password**.

9. Ensure **Required to register** is checked.

10. Click **Next**.

11. Verify the information in the **Summary** section and click **Finish**.

You should be able to receive calls at your telephone number after finishing these steps. You should also be able to call numbers within the country in which you have chosen your phone number.

However, you have to do a few additional steps to be able to call international numbers.

Skype requires that you send international numbers in the E.164 format with + in front of the country code. In our example, the dial-out prefix is `9` and we will add the prefix `900` for international calls. The approach is as follows:

1. Go to the **Call Routing** screen again and click **Add** in **Routing of outgoing calls**.

2. In the **Add Outgoing Route** dialog, enter `900` as the route prefix and add your Skype Connect interface as created before.

3. Modify the values for the **Called number** so that 3 digits are stripped off as we want to remove the prefix `900`) and replace is with `+`.

4. Click **OK** to save the route.

Now, as an example, if you want to call a number in the Netherlands, you would dial `90031`...and this would get translated this to `+31`... and will send the call to your Skype Connect interface.

The Skype™ name, Skype Connect™ and their logos are trademarks of Skype.

2.8 Gateways

This section helps you connect via various gateways.

2.8.1 Configuring Kerio Operator and Cisco SPA8800 for calls over an analog telephone line

Cisco SPA8800 is an analog-to-SIP gateway equipped with 4 FXS and 4 FXO telephone ports.

> **NOTE**
> This information is designed for Kerio Operator 2.4 and newer.

Learn how to configure the SPA8800 with Kerio Operator to place and receive phone calls over the FXO interface (analog telephone line).

Prerequisites

We assume that your Kerio Operator is up and running and you have at least one internal extension.

We also expect:

» Your SPA8800 is already connected to your LAN and you have access to its web administration. To discover the device's IP address or change the basic network settings, you can connect an analog phone to the **Phone 1** port on the SPA and use the device's simple IVR system as described in the SPA8800 Quick Start Guide.

» Your PSTN line is connected to the **Line 1** port of the SPA device.

Additionally, we have describe the configuration of a single line. If you are about to use several lines on the SPA, you need to repeat the configuration steps up to 4 times. The SPA8800 uses a separate SIP/UDP port for every analog line, so you need one SIP interface in Kerio Operator for each line in use. The default port numbers are `5061` for Line 1, `5161` for Line 2, 5261 for Line 3, and `5361` for Line 4.

Configuring Kerio Operator

> **NOTE**
> The PSTN phone number `5550199`, Kerio Operator IP address `10.1.2.95` and SPA8800 IP address 10.1.2.200 are all sample values and are used as examples to explain the configuration process.

1. In the administration interface, go to **Call Routing**.

2. Create a new SIP interface with `5550199` as the external number.

3. Click **Next**.

4. In the second screen, choose an extension to receive calls from this interface and set a dial-out prefix (for example 9).

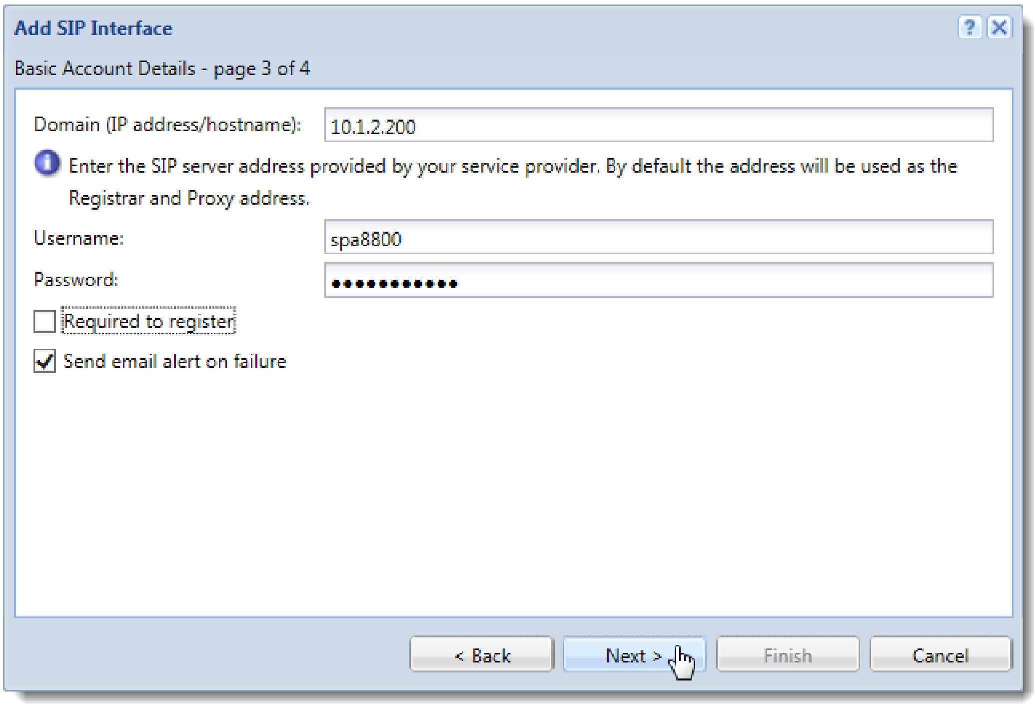
5. In the third screen, set the following:

a. Set the IP address of the SPA device. For example, `10.1.2.200`

b. Set the username. Kerio Operator uses to authenticate with the SPA. For example, `spa8800`.

c. Set a password, for example `pass5550199`.

d. Clear the **Required to register** option. The SPA is not able to behave as a SIP registrar.



For more information, refer to Connecting Kerio Operator to TelePacific (page 52).

6. In the last screen, select **Edit details of the created interface** and click **Finish**.

After you finish the configuration of the SIP interface, the **Edit External Interface** dialog box opens:

1. Go to the **SIP Details** tab.

2. In the **Outbound Proxy** field, set the IP address of the SPA device with a port number 5061. For example, `10.1.2.200:5061`.

3. Leave the **Inbound Proxy** field empty.

4. Click **OK** to save your changes.

Configuring SPA8800

1. Connect to the web interface of your SPA device. For example, `10.1.2.200`.

2. Go to the administration section (log in as the administrator, if needed) and click **Voice** and then **Advanced** — in our example, we reach the URL http://10.1.2.200/admin/voice/advanced.

3. Go to the **Line 1** configuration page. All subsequent steps (except the step 11) will be done on this page.

4. Verify that **Line Enable** is set to **yes**.

5. Under SIP Settings, verify that **SIP Port** is set to **5061**. Set the **Auth INVITE** field to **yes**.

6. Scroll down to the **Proxy and Registration** section:

a. **Proxy** contains Kerio Operator's IP address. For example, `10.1.2.95`.

b. Set **Use Outbound Proxy** to **no**.

c. Set **Register** to **no**.

d. Set **Use OB Proxy In Dialog** to **no**.

e. Set **Make Call Without Reg** to **yes**.

f. Set **Ans Call Without Reg** to **yes**.

5. Scroll to the **Subscriber Information** section:

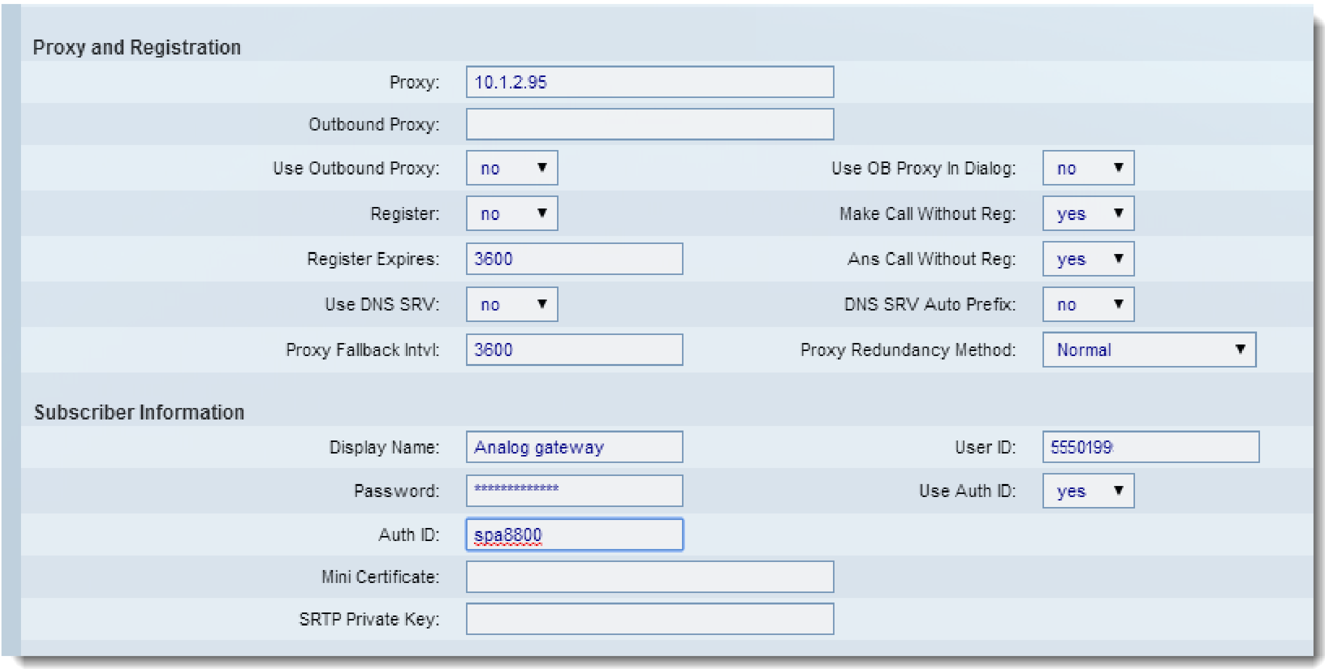
a. Set **Display Name** to some non-empty string. Use something like External call or Analog device to make it easier for your users to distinguish calls coming through the SPA device.

b. Set **User ID** to your external telephone number, for example `5550199`.

c. Set **Password** to your chosen authentication password, for example `pass5550199`.

d. Set **Use Auth ID** to **yes**.

e. Set **Auth ID** to your chosen authentication user name. For example, `spa8800`.



Screenshot 5: Configuring Proxy and Subscriber Information in SPA8800

6. Scroll to the **Dial Plans** section. Set **Dial Plan 8** to `S0<:5550199@10.1.2.95>`. Here, `5550199` is the external phone number and `10.1.2.95` is the IP address of Kerio Operator.

7. Now move to the VoIP-To-PSTN Gateway Setup section. Set the **VoIP Caller Default DP** field to **none**.

8. Scroll to the **PSTN-To-VoIP Gateway Setup** section:

a. **PSTN CID Number Prefix** should be empty.

b. **PSTN Caller Default DP** must be **8**. (DP stands for Dial Plan — see step 6 above).

> **NOTE**
> Note about Caller ID: On a typical PSTN line, the Caller ID (aka CLIP or CID) is usually transmitted using the FSK modulation between the first and the second ring of the incoming call. If your PSTN line provides the Caller ID service, set **PSTN CID For VoIP CID** to **yes** but also increase the value in **PSTN Answer Delay** (under **FXO Timer Values**) so that the SPA device can hear the 2 rings before starting the VoIP call to Kerio Operator.

9. Go to the **FXO Timer Values (sec)** section:

    a. Set **VoIP Answer Delay** to **0**.

    b. Set **PSTN Answer Delay** to **0**. If you want to transfer the PSTN Caller ID to the VoIP side, enter a number (in seconds) that approximately corresponds to 2 rings on your PSTN line.

10. Save the configuration by clicking on **Submit All Changes** at the bottom of the configuration page. Note that the device might need 20-30 seconds to save and apply the new configuration.

11. If you enabled the Caller ID detection in step 8, go to the **Regional > Miscellaneous** and check that the fields **Caller ID Method** and **Caller ID FSK Standard** correspond with the standard used by the local telco company. The most usual method is **Bellcore — bell202**, but you may need to ask the telco about the Caller ID encoding method they are using.

Both Operator and the SPA8800 are now configured and you can place some test calls. If you are not satisfied with the volume of the sound transmitted from the analog side to VoIP or vice versa, you can return to the **Line 1** configuration page, scroll to **International Control** and change the values for **SPA To PSTN Gain** and/or **PSTN To SPA Gain**.

2.8.2 Configuring Kerio Operator and Grandstream GXW 4104/4108 for calls over analog telephone lines

Grandstream GXW 4104 is an analog-to-SIP gateway with four FXO ports. The GXW 4108 model has eight FXO ports.

Prerequisites

To complete the configuration, you need:

- Kerio Operator up and running.
- At least one internal extension.
- The Grandstream gateway connected to your LAN.
- One analog phone connected to the first port of the GXW gateway.

In the example below:

- The Kerio Operator IP address is 10.1.2.95.
- The Grandstream gateway's IP address is 10.1.2.200.
- The phone number of connected phone is 1234567.
- The internal extension's number is 100.

> **NOTE**
> The example below uses GXW 4104, firmware version 1.4.1.5. The GXW gateway runs without a SIP password in a peer-to-peer configuration. Use this configuration in a safe local network.

Kerio Operator configuration

1. In the administration interface, go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.

2. Click **Add SIP Interface**.

3. Type a name for the interface (for example, the provider's name). The name must not contain spaces or special characters and must be unique.

4. Select **New provider**.

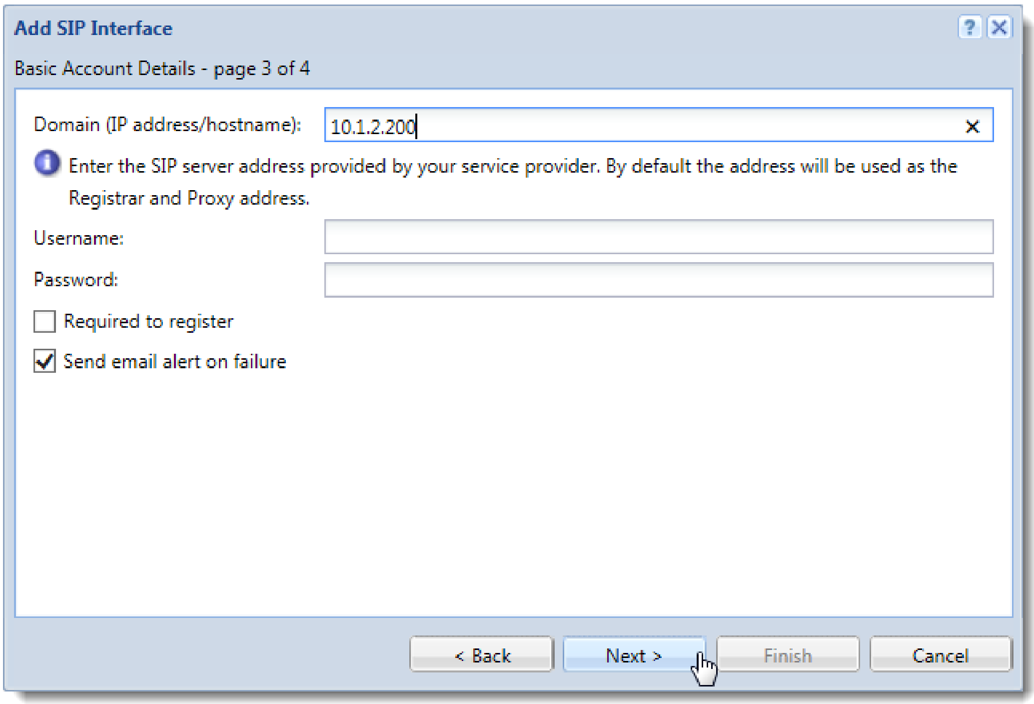
5. In the **With external number** field, type the number `1234567` and click **Next**.

6. Select an extension that receives all calls, For example, (`100`.

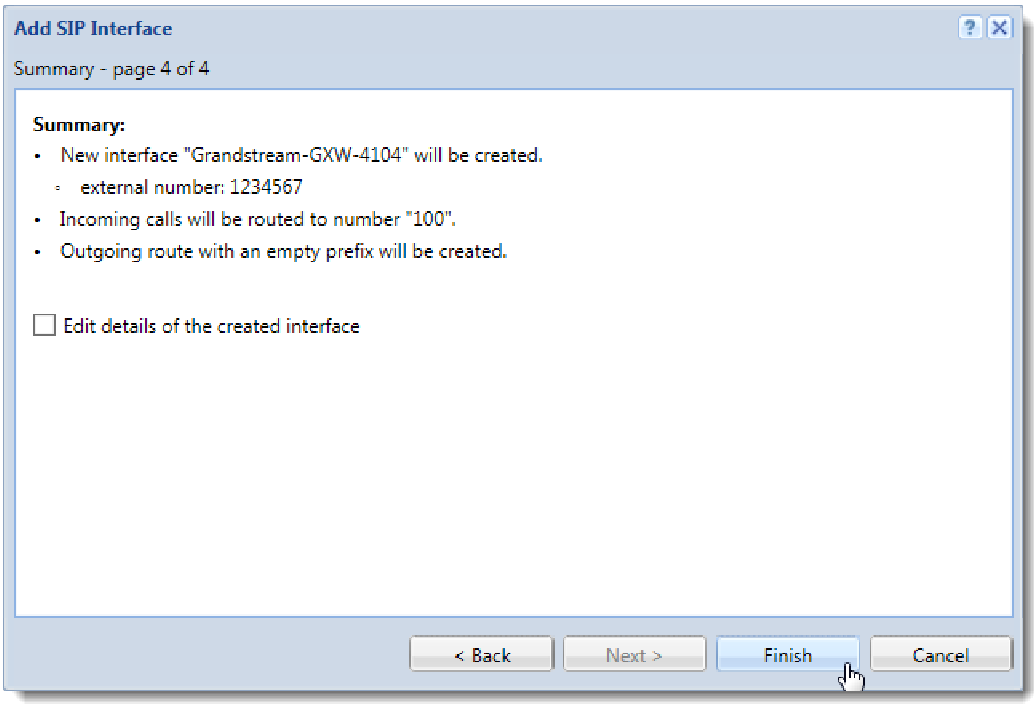
7. Optionally, in the **Prefix to dial out** field, type a prefix for outgoing calls and click **Next**.

8. In the **Domain (IP address/hostname)** field, type the IP address of the Grandstream GXW gateway, `10.1.2.200`.

9. Deselect the **Required to register** option and click **Next**.



10. Verify the information in the **Summary** section and click **Finish**.



Grandstream GXW configuration

To configure the GXW 4104 model, use the `ch1-4` parameter. For a GXW 4108 model configuration, use the parameter`ch1-8`.

1. In the Grandstream GXW administration interface, go to **Accounts > Account 1 > General Settings**.

2. In the **SIP Server** field, type the IP address of (`10.1.2.95`).

3. In the **Account Active** field, select **Yes**.

4. Optionally, set up your **Account Name** and **Outbound Proxy**.

5. Save your settings.

The web interface now asks whether you want to reboot to apply the new configuration. You can reboot now or continue with the configuration and reboot later.

1. Go to **Accounts > User Account** and verify that the configuration form is empty.

2. Go to **Settings > Channel Settings > Calling to VoIP**.

   a. Set **User ID** to `ch1-4:1234567;`. Calls that come to any of the four analog ports are send to Kerio Operator as if the number called was 1234567. To differentiate among the analog ports, type additional external numbers (for example, `1234568`) in the SIP interface in Kerio Operator and set **User ID** to `ch1-2:1234567;ch3-4:1234568`. This configuration maps analog ports 1 and 2 to the number `1234567` and ports 3 and 4 to the number `1234568`.

   b. In the **SIP Server** field, set `ch1-4:pl;`.

   c. In the **SIP Destination Port** field, set `ch1-4:5060;`.

3. Go to **FXO Lines > Settings > Port Caller ID Setting**.

   a. In the **Number of Rings Before Pickup** field, set `ch1-4:2;`. Analog lines usually transmit the caller ID between the first and second ring. If you want to detect the caller ID, you must allow at least two rings before

accepting the call.

b. In the **Caller ID Scheme** field, set the option to the method used by your analog line provider. For example, for Bellcore use `ch1-4:1;`.

c. In the **Caller ID Transport Type** field, set `ch1-4:1;` to transmit the caller ID in the SIP **From** field.

4. Go to **FXO Lines > Dialing > Dialing to PSTN**, and in the **Stage Method (1/2)** field, set `ch1-4:1;`.

5. Reboot the gateway.

After the reboot, make several test calls to verify the configuration.

> **NOTE**
> To change the volume of the sound transmitted between the analog line and the SIP, go to **FXO Lines > Settings > Port Voice Settings** and modify the audio gain values.

2.8.3 Configuring Kerio Operator and Grandstream GXW4224 to use analog phones for internal extensions

Grandstream GXW4224 is an analog-to-SIP gateway that allows you to connect up to 24 analog phones. Similar models with 16, 32, and 48 FXS ports exist.

Prerequisites

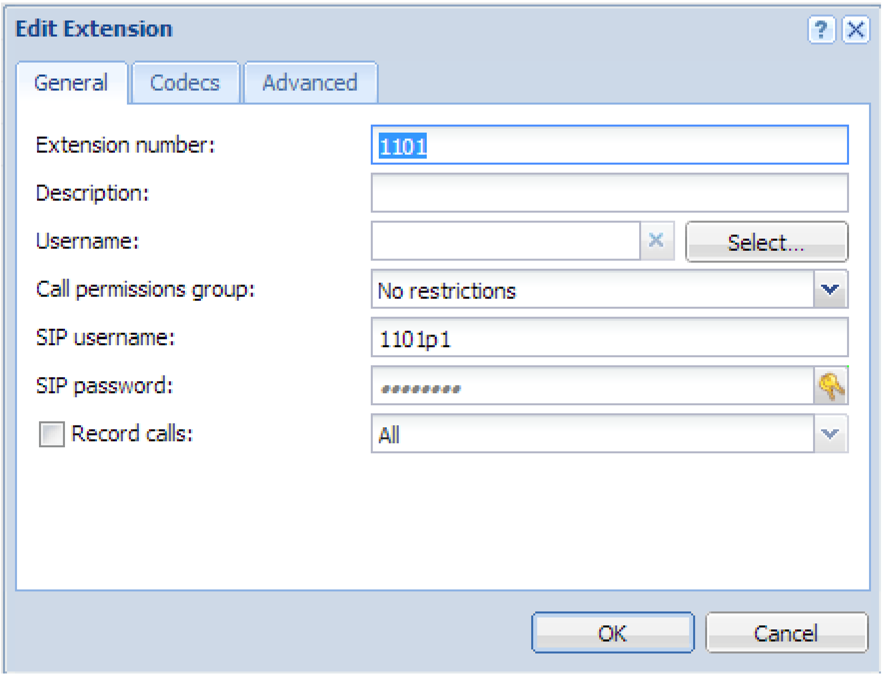
Before going into actual process, we assume the following:

- Kerio Operator is up and running.
- You have at least one other SIP phone that allows you to place a test call.
- Grandstream gateway is connected to your LAN and you have access to its web configuration interface.
- At least one analog phone is connected to the Grandstream gateway (to the 1st FXS port);

Kerio Operator configuration

1. In the administration interface, go to **Extensions**.

2. Create the extensions you are about to assign to the analog phones. In this example, we create just a single extension `1101`.

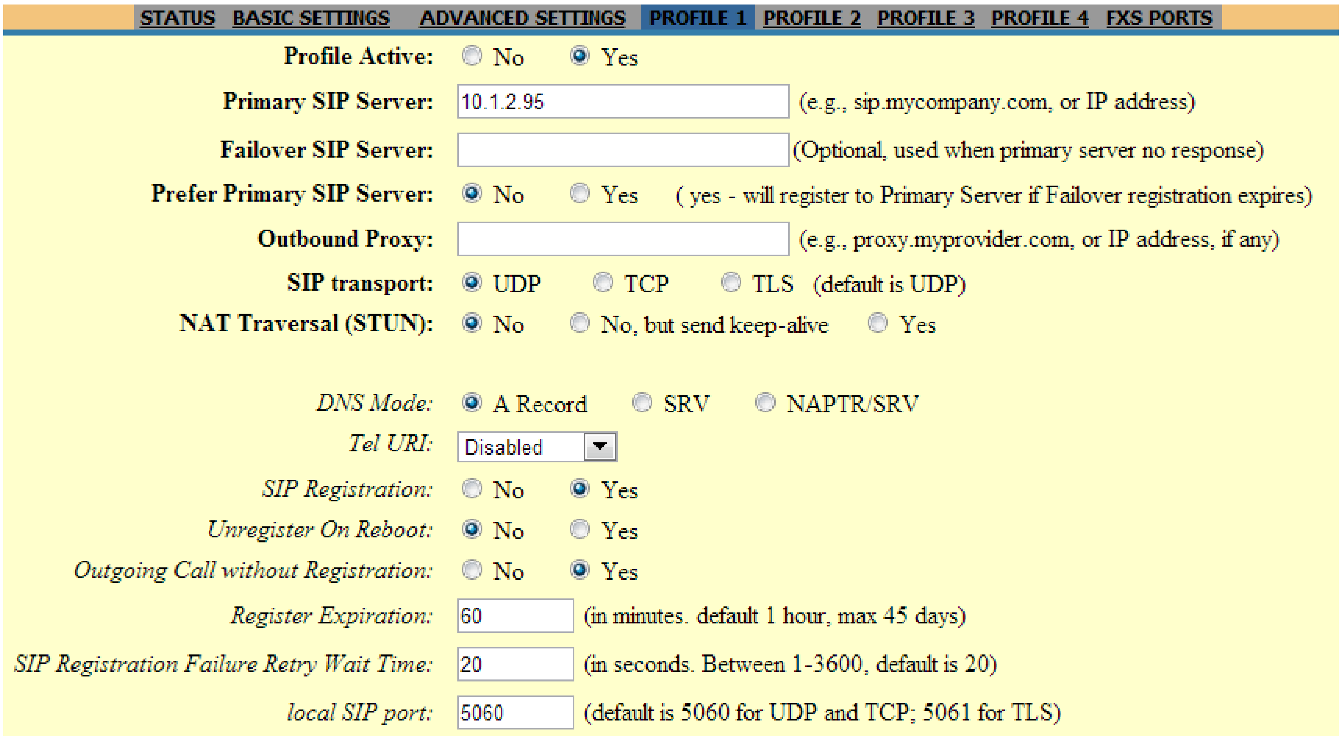


Screenshot 6: Edit Extension dialog

NOTE
The SIP username can be the same as extension number but we use the SIP username `1101p1` to show that the username can be different. Also, write down the SIP password or copy it to the clipboard.

Grandstream GXW4224 Configuration

1. Connect to the Grandstream's web interface.
2. Go to **Profile 1** (see screenshot below). The Grandstream gateway lets you configure up to 4 SIP servers.
3. Configure Kerio Operator as the 1st server (Profile 1).
4. In the Profile 1, check that **Profile Active** is set to **Yes**.
5. Key in Kerio Operator's IP address as the **Primary SIP Server**. For example, `10.1.2.95`.
6. Check that the field **DNS Mode** is set to the value **A Record**.
7. Check that the **Local SIP port** is set to `5060`.



Screenshot 7: Profile 1 screen

8. Scroll down to the bottom of the **Profile 1** page and click **Update**.

9. Go to the screen **FXS Ports**.

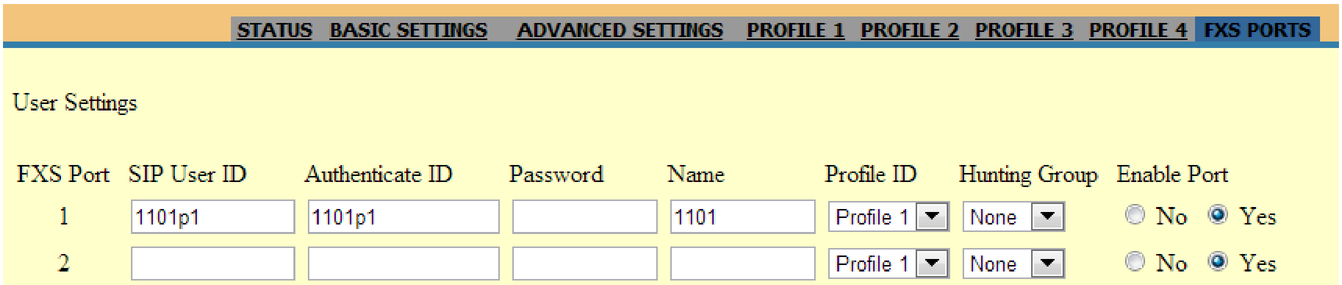
10. Key in the SIP identifiers, password, and server profile for each of the 24 analog ports (or 16, 32, 48 ports, respectively). To configure the extension 1101, we do the following for **FXS Port** 1:

   a. Enter `1101p1` into both **SIP User ID** and **Authenticate ID** fields.

   b. Enter the extension's password.

   c. Enter `1101` into the **Name** field.

   d. Select **Profile 1**.



Screenshot 8: FXS Ports

11. Scroll down to the bottom of the screen and click **Update**.

12. Reboot the gateway.

Testing

When the reboot is complete, you should see the extension registered in Kerio Operator's **Extensions** grid. Also the Grandstream's **Status** screen should show the registered extensions. You can now place some test calls between analog

phones connected to the gateway and your other SIP phones.

2.8.4 Configuring Kerio Operator and Grandstream HT503 for calls over analog lines

The Grandstream HT503 is an analog-to-SIP ATA device with one FXO and one FXS port. You can use this device with Kerio Operator to make calls over the telephone network.

Prerequisites

To complete the configuration, you need:

- Kerio Operator up and running.
- At least one internal extension.
- Telephone line connected to the FXO port.
- The Grandstream HT503 connected to the same LAN as Kerio Operator.

The example in this topic uses the following inputs:

- The Kerio Operator IP address is `10.1.2.95`.
- The Grandstream HT503's IP address is `10.1.2.200`.
- The external number from the provider is `123456`.
- The SIP password used for the FXO configuration on the Grandstream HT503 is `pass1234`.

Connecting to a network with another DHCP server

The Grandstream HT503 runs a DHCP server on the LAN port. If you have another DHCP server in your network, connect the device to your network via the WAN port and enable web access to the administration interface on that port.

To enable the access:

1. After the device boots, connect an analog phone to the FXS port.
2. Press `***` to access the voice menu.
3. Press `12` and then `9`.

To hear the IP address of the device, press `***` and then `02`.

Configuring the Grandstream HT503

To configure the Grandstream device, start by configuring the FXO port:

1. In the administration interface of the device, go to the **FXO PORT** section.
2. In the **Primary SIP Server** field, type the Kerio Operator IP address. For example, `10.1.2.95`.
3. In the **SIP User ID** and **Authenticate ID** fields, type the external number. For example, `123456`.
4. In the **SIP Registration** field, select **No**.
5. Optionally, in the **Caller ID Scheme** field, select the method used by your provider to detect the caller's ID for incoming calls.
6. In the **Caller ID Transport Type** field, select **Relay via SIP From**.
7. In the **Number of Rings** field:

- To detect the caller's ID for incoming calls, type `2`.
- If your provider does not offer Caller ID, type `1`.

8. In the **PSTN Ring Thru FXS** field, select **No**.

9. In the **Wait for Dial-Tone** field, select **No**.

10. In the **Stage Method** field, type `1`.

11. Click **Update** and then **Apply** to save your changes.

After you finish configuring the FXO port, configure the rest of the settings:

1. Go to the **BASIC SETTINGS** section.

2. Go to the **Unconditional Call Forward to VOIP** section.

3. In the **User ID** field, type the external number (`123456` in the example).

4. In the **SIP Server** field, type the Kerio Operator IP address (`10.1.2.95` in the example).

5. In the **SIP Destination Port** field, type `5060`.

6. Click **Update** and then **Apply** to save the changes.

Configuring Kerio Operator

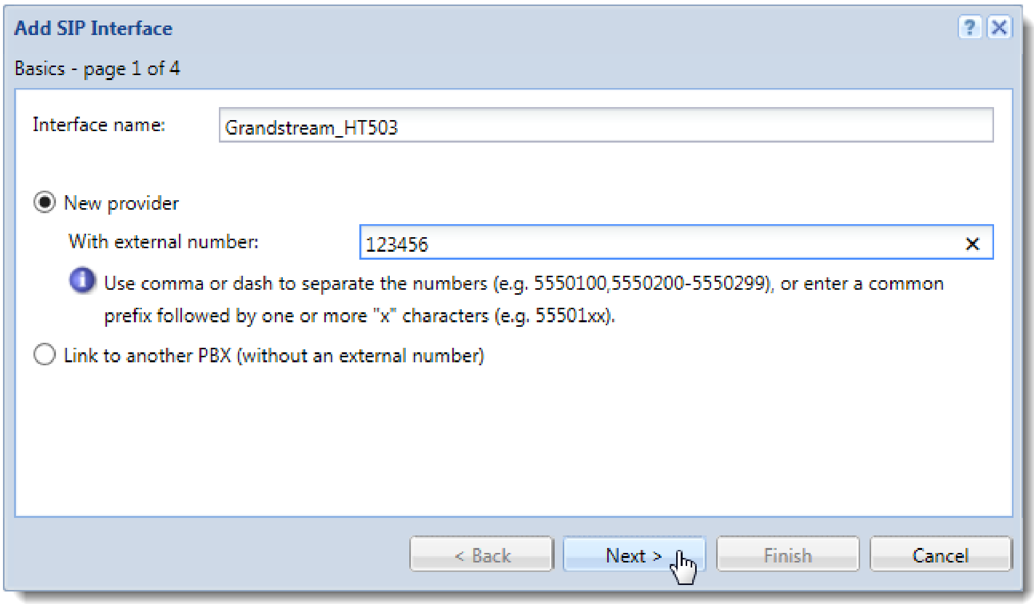
1. In the administration interface of Kerio Operator, go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.

2. Click **Add SIP Interface**.

3. Type a name for the interface (for example, the name of the gateway). The name must not contain spaces or special characters and must be unique.

4. Select **New provider**.

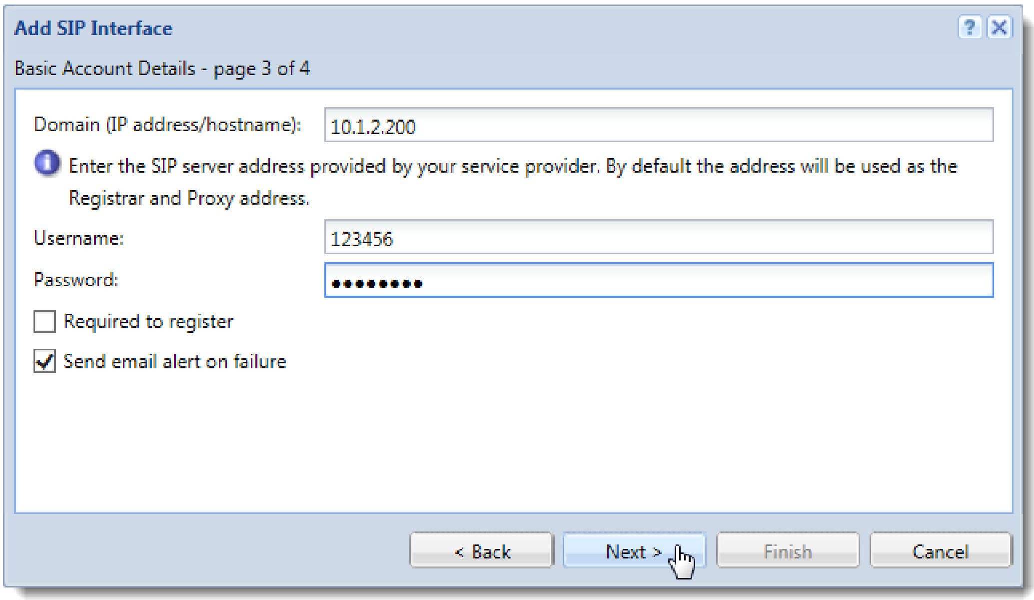
5. In the **With external number** field, type the external number (`123456`) and click **Next**.



6. Select an extension that receives all calls.

7. Optionally, in the **Prefix to dial out** field, type a prefix for outgoing calls and click **Next**.

8. In the **Domain (IP address/hostname)** field, type the IP address of the Grandstream device (`10.1.2.200`).



9. In the **Username** field, type the external number.

10. In the **Password** field, type the SIP password used for the FXO configuration on the Grandstream HT503. For example, `pass1234`.

11. Clear the **Required to register** option and click **Next**.

12. Select the **Edit details of the created interface** option and click **Finish**.

After Kerio Operator finishes the configuration wizard, the Edit External Interface (SIP) dialog box opens:

1. Go to the **SIP Details** tab.

2. In the **Outbound proxy** field, type `10.1.2.200:5062`.

3. In the **Inbound proxy** field, type `10.1.2.200:5062`.

4. Click **OK** to save your changes.

After you complete the configuration, make some test calls to verify the connection between Kerio Operator and Grandstream HT503.

2.8.5 Configuring Kerio Operator and WellTech 2504/WellGate 2504 to use analog phones for internal extensions

WellTech 2504 is an analog-to-SIP gateway with 2 Ethernet ports and 4 FXS ports that allow you to connect up to 4 analog phones. It is sold as WellGate 2504 in some markets.

Learn how to integrate Kerio Operator with WellTech 2504/WellGate 2504.

Prerequisites

Before starting the configuration, we assume:

- Your Kerio Operator is up and running. In the example below the Kerio Operator's IP address has been used as `10.1.2.95`.
- You have at least one other SIP phone that allows you to place a test call.

» Your WellTech 2504 is connected to your LAN and you have access to its web configuration interface.

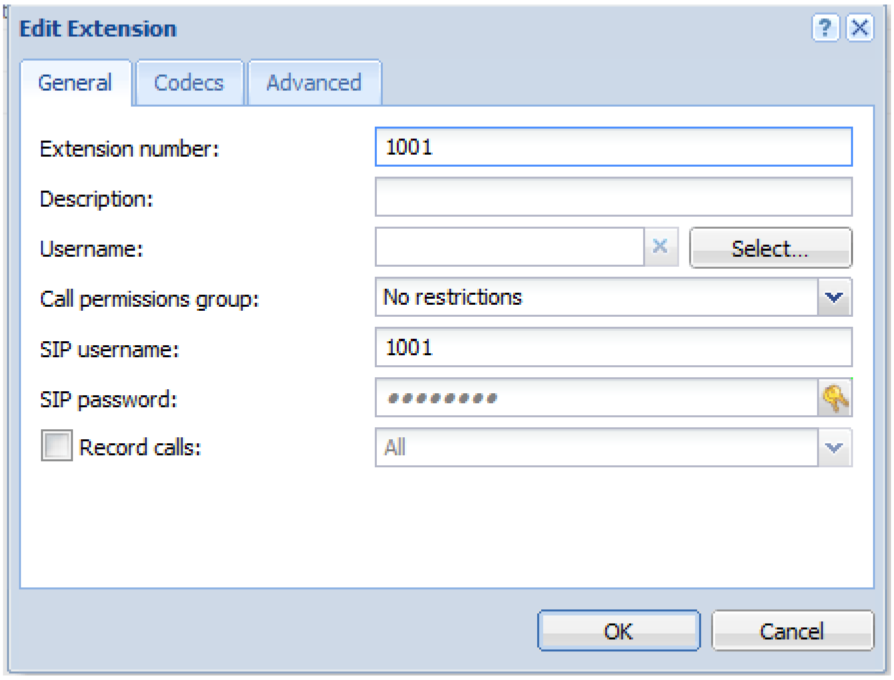
» At least one analog phone is connected to WellTech 2504 (to port Tel 1).

NOTE
The WellTech gateway runs a DHCP server on its LAN interface by default. If you already have a DHCP server running in your LAN, do not connect the gateway's LAN interface to your network straight away. Following the device's quick installation guide, you can first connect a single computer to the WellTech's LAN interface and set up networking as needed for your LAN.

Kerio Operator configuration

1. In the administration interface, go to **Extensions**.

2. Create the extensions you are about to assign to the analog phones. In this example, we create just a single extension 1001.



Screenshot 9: Edit Extension dialog

NOTE
Because of the limitations in how the WellTech gateway handles SIP identifiers, the SIP username must be the same as the extension number (1001). Also. take a note of SIP password or copy it to the clipboard.

WellTech 2504 Configuration

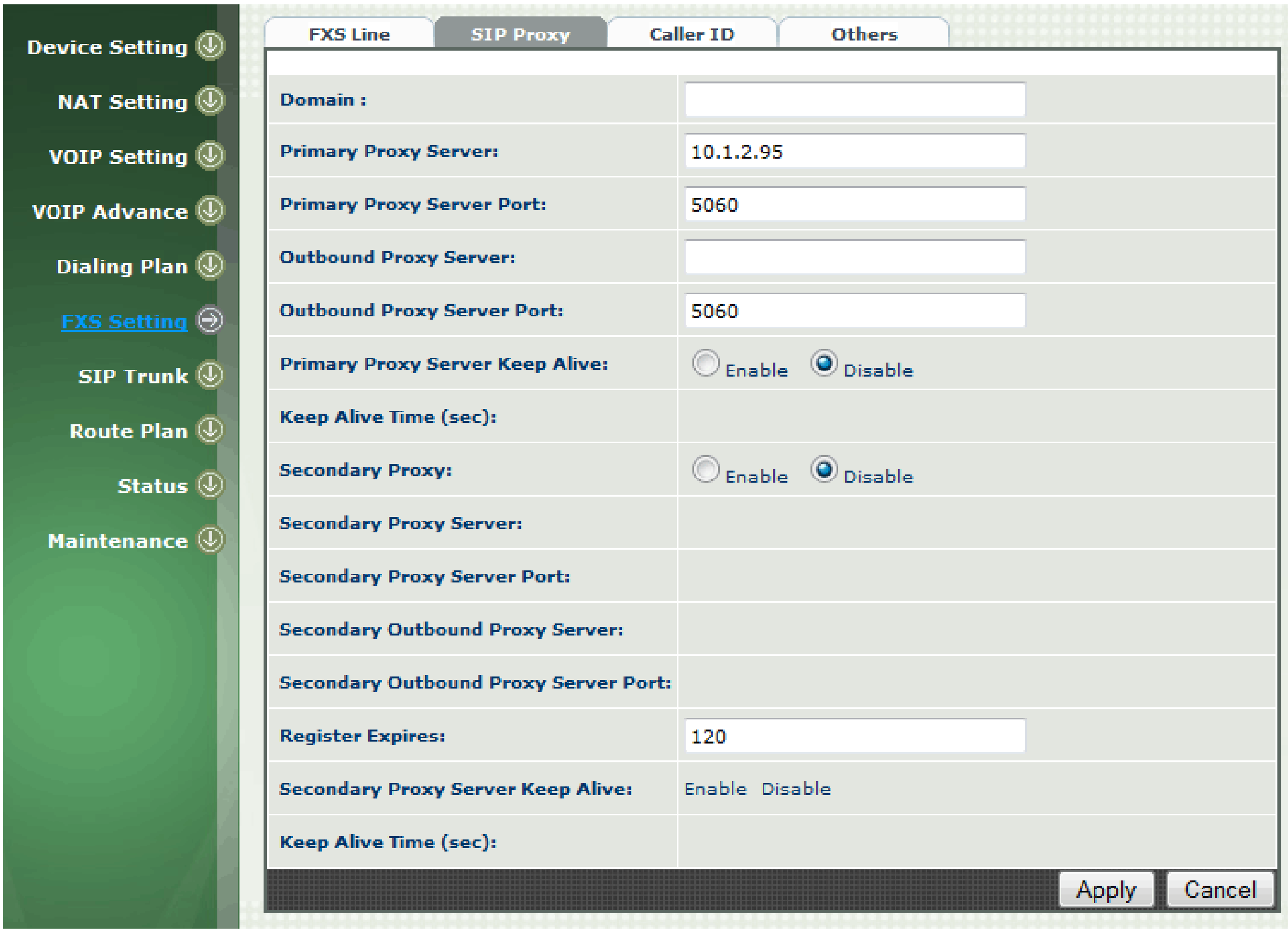
1. Connect to the WellTech's web interface.

2. Go to **FXS Settings > SIP Proxy**.

3. In the **Primary Proxy Server**, type Kerio Operator's IP address.

4. Check that the **Primary Proxy Server Port** is set to 5060.

5. Click **Apply**.



Screenshot 10: SIP Proxy screen

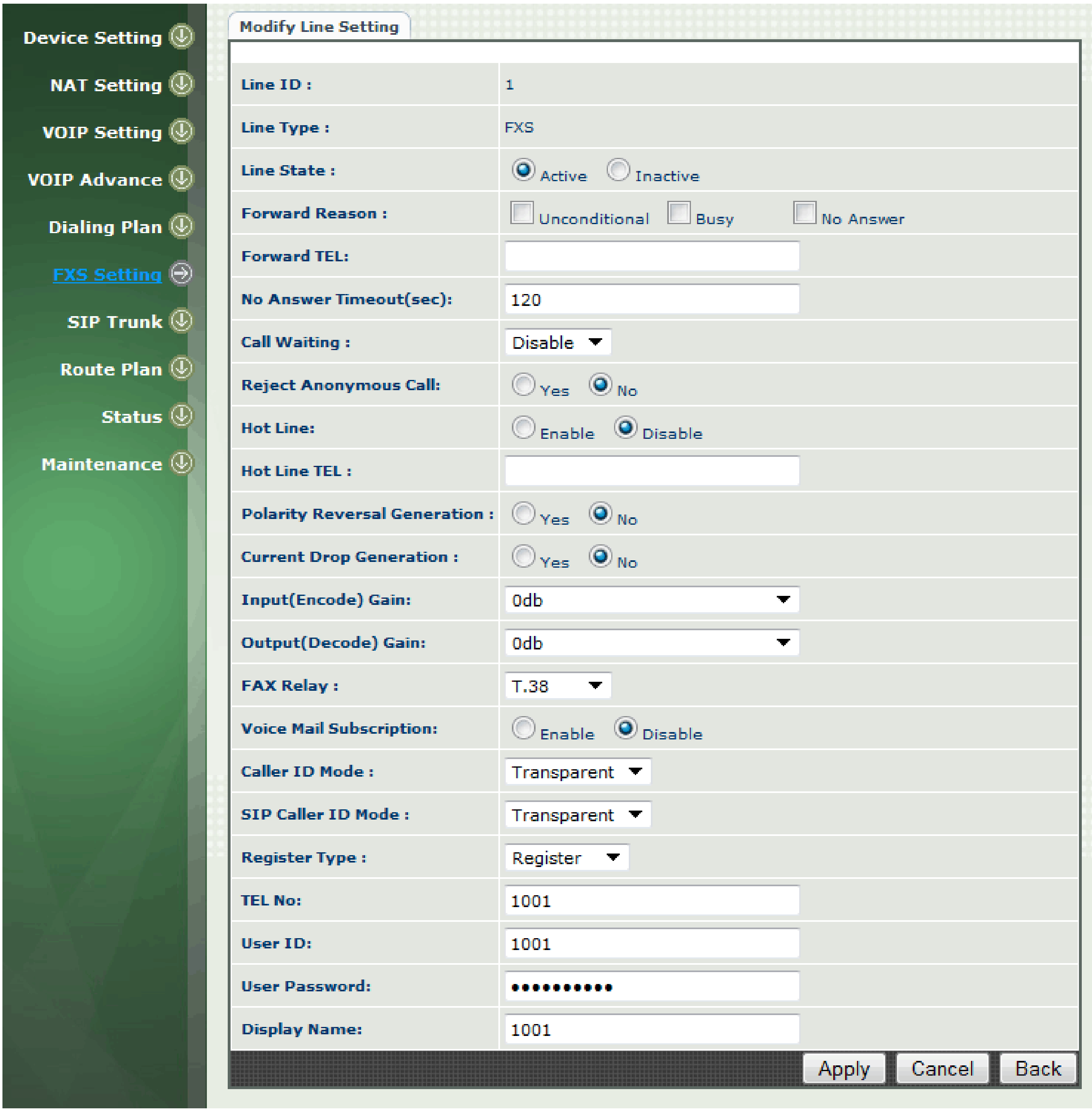
6. Go to **FXS Settings > FXS Line** and click the edit icon for the first line. The edit icons are in the first column of the table. The line edit screen should appear.

7. Scroll to the bottom of the screen, set the field **Register Type** to **Register** and type the extension number (1001) into the **TEL No** field.

8. Type the extension's SIP username into the field **User ID** (it must be the same as the extension number 1001).

9. Type (or paste) the extension's SIP password.

10. Click **Apply**



Screenshot 11: FXS Setting

If you are using multiple lines, configure them in a similar way. The unused lines should be set as inactive (the field **Line State**).

The WellTech gateway needs to be restarted to start using the new configuration. Go to **Maintenance** and reboot the device.

Testing

The WellTech gateway may need about a minute to reboot. As soon as it is up and running, the extensions should appear as registered in Kerio Operator's **Extensions** grid. You can now place some test calls between the analog phones and another SIP phone.

2.8.6 Configuring Kerio Operator and Well/Yeastar NeoGate TB400 for calls between SIP and EuroISDN

NOTE
This information is designed for Kerio Operator 2.3.5 and older.
For more information about creating and configuring a SIP interface in newer versions, see Connecting to VoIP service providers.

Yeastar NeoGate TB400 (sold under the Well brand in some countries) is a SIP-to-ISDN gateway. The gateway can be equipped with 2 or 4 BRI ports and hence it supports up to 8 parallel calls.

Prerequisites

Before starting the configuration, we assume:

- Kerio Operator is up and running.
- You have at least one other SIP phone that allows you to place a test call.
- Your NeoGate TB400 is connected to your LAN and you have access to its web configuration interface.
- You have one EuroISDN line connected to the 1st BRI port on the NeoGate.
- You have two external phone numbers assigned to your EuroISDN line.
- You know the signaling type for your ISDN line (point-to-point or point-to-multipoint).

In the example below:

- The Kerio Operator IP address is `10.1.2.95`.
- The NeoGate TB400 IP address is 10.1.2.200.
- The two external phone numbers are `300123456` and `300123457`.

NeoGate TB400 configuration

Connect to the web administration interface of your TB400 and do the following steps:

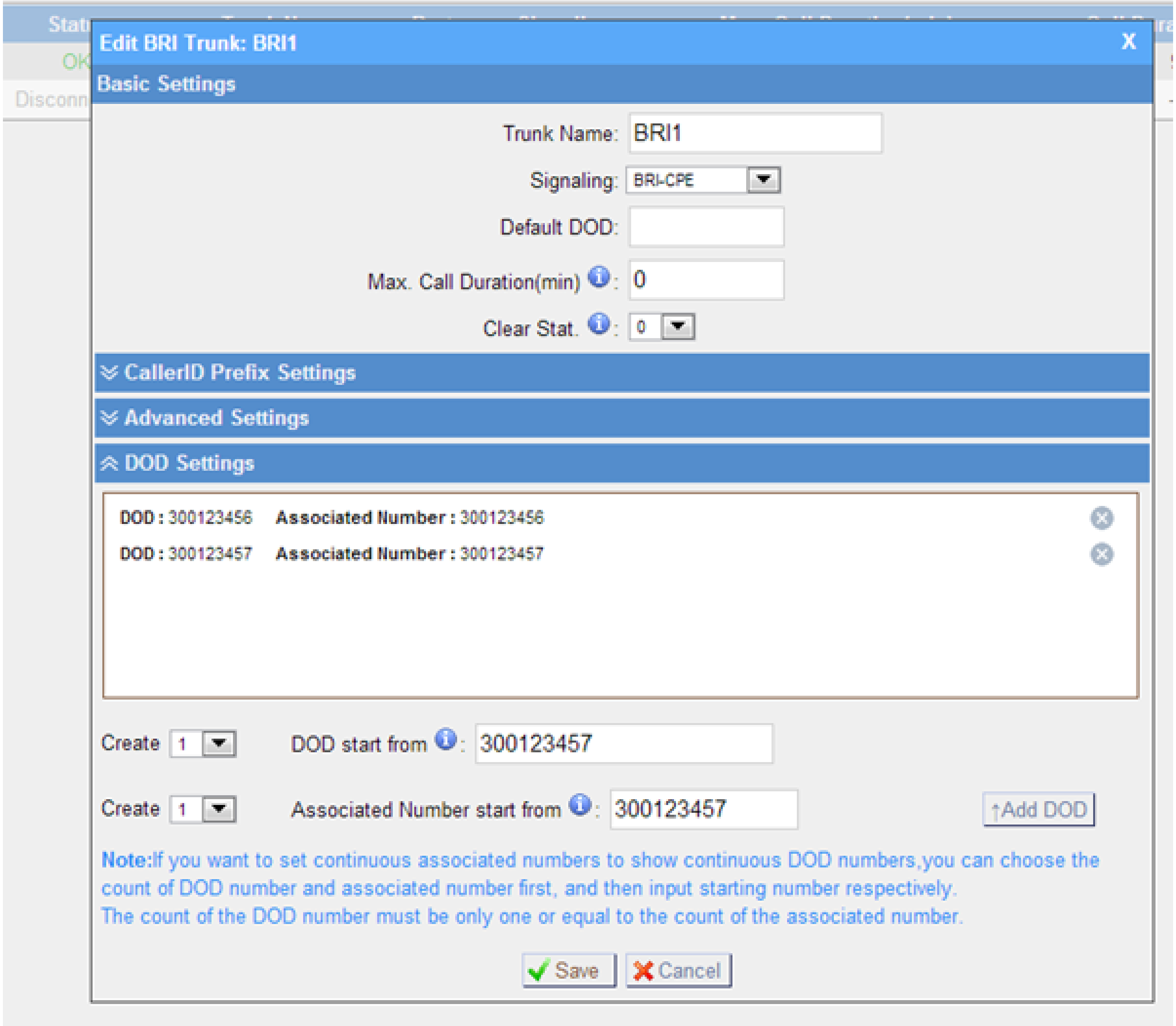
1. Go to the **BRI Settings > Module List** screen and open the edit dialog for module BRI1.

2. Set **Signaling** to `BRI-CPE` or `BRI-CPE-PTMP` depending on whether your line uses point-to-point or point to multi-point signaling, respectively.

3. Switch to the **DOD Settings** section in the edit dialog and key in the first external phone number (`300123456`) in both **the DOD start from** and **Associated number start from** fields.

4. Click **Add DOD** and add your second external number (`300123457`) as done in the previous step.

5. Click **Save** to close the edit dialog.



Screenshot 12: Configuring external numbers

6. Go to the **SIP Settings > Trunk** screen.

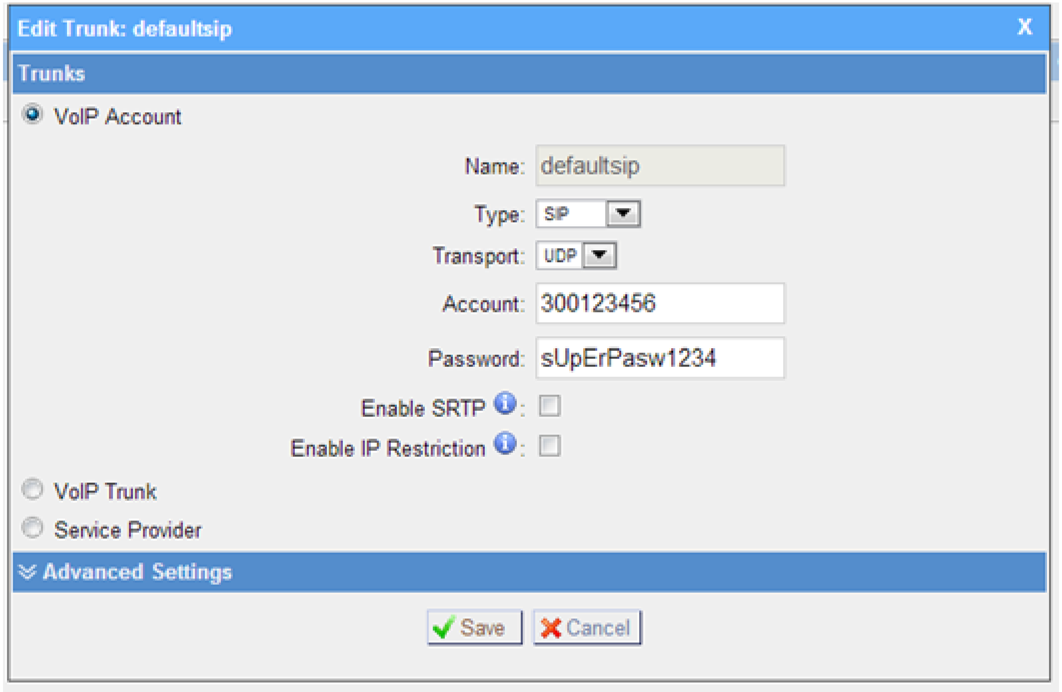
7. Select **VoIP Account**.

8. Set the **Type** field to `SIP` and **Transport** to `UDP`.

9. Type your first external phone number (`300123456`) into the **Account** field.

10. Type some complex password into the **Password** field.

11. Click **Save**.



Screenshot 13: Configuring VoIP account in TB400

12. Click **Apply Changes**.

> **IMPORTANT**
> You may encounter a situation with gateway not using the DOD numbers on the BRI interface correctly until restarted. So you may also want to restart NeoGate at this point

Kerio Operator configuration

1. Connect to your Kerio Operator administration GUI and go to the **Call Routing** screen.

2. Create a new SIP interface. Enter your external numbers separated with comma in the first screen of the **Add SIP Interface** wizard. For example, `300123456,300123457`.

3. In the second screen of the **Add SIP Interface** wizard, select an extension to receive calls from this interface and set a dial-out prefix (in our example, let's use 9 as the prefix).

4. Enter the following values in the third screen:

    a. Key in the gateway's IP address (`10.1.2.200`) in the **Hostname or IP address** field.

    b. **Port number** should be `5060`.

    c. The **Username** should be your phone number (`300123456`).

    d. Key in your TB400 VoIP account password.

    e. **Must register with the Registrar or Proxy** should remain checked.

You can now place some test calls. Use the prefix you have assigned to the new SIP interface to call via NeoGate TB400.

2.8.7 Configuring Kerio Operator and Well/Yeastar NeoGate TG200 for calls between SIP and GSM

Yeastar NeoGate TG200 (Well NeoGate TG200 in some countries) is a SIP-to-GSM gateway. The gateway can be equipped with one or two GSM modules and supports at most two parallel calls.

NOTE
This information is designed for Kerio Operator 2.4 and newer.

Prerequisites

To complete the configuration, you need:

» Kerio Operator up and running.

» At least one other SIP phone that allows you to place a test call.

» The NeoGate TG200 gateway connected to your LAN.

» At least one SIM card inserted into NeoGate TG200 and the SIM card's PIN number in the NeoGate's web configuration.

In the example below:

» The Kerio Operator IP address is 10.1.2.95.

» The NeoGate gateway IP address is 10.1.2.200.

» The SIM card number is 700123456.

Configuring NeoGate TG200

Connect to the web administration interface of your NeoGate TG200:

1. Go to **SIP Settings > Trunk**

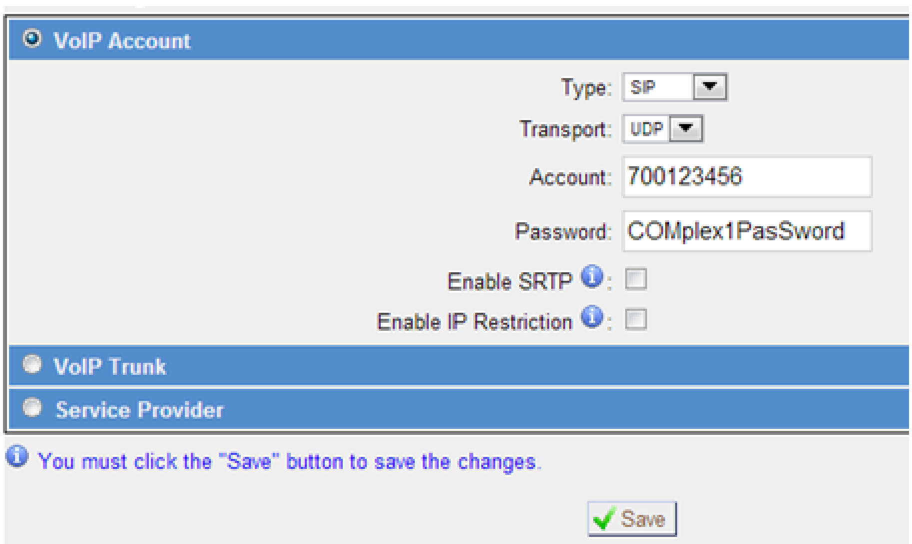
2. Select the **VoIP Account** option.

3. Set the **Type** field to **SIP** and **Transport** to **UDP**.

4. In the **Account** field, key in the SIM card number.

5. In the **Password** field, key in a password. Use this password when configuring the SIP interface.

6. Click **Save**.



Screenshot 14: Configuring VoIP account in TG200

7. Go to **Route Settings > Outgoing Routes**.

8. Add a new outgoing route.

> **NOTE**
> The default route in NeoGate TG200 does not work when calling numbers in the international format with "+" at the beginning.

If you need the international format when calling back, add another route as a workaround:

1. Go to **Route Settings > Outgoing Routes**.

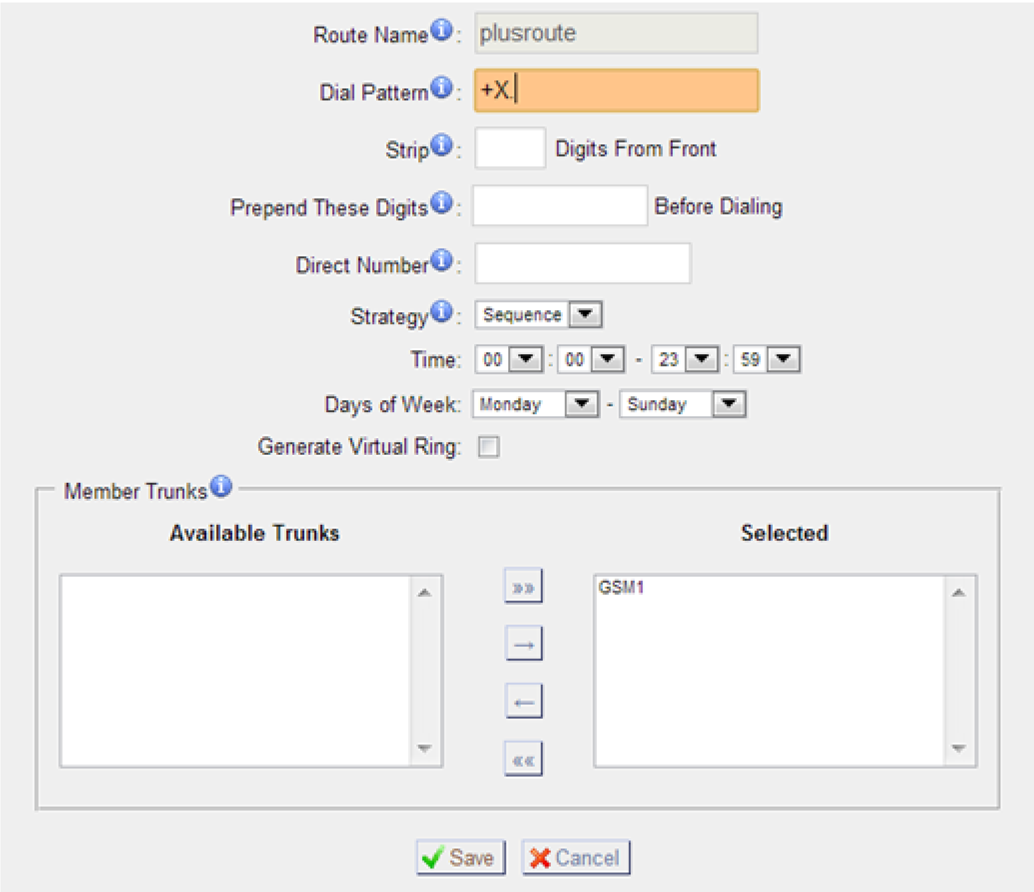
2. Add a new outgoing route.

3. In the **New Outgoing Route** dialog box, key in a new route name in the **Route Name** field.

4. In the **Dial Pattern** field, key in the `+X.` string.

5. Move at least one GSM module from **Available Trunks** to **Selected**.

6. Click **Save**.



Screenshot 15: The route to handle the international number format in TG200

Configuring Kerio Operator

1. In the administration interface of Kerio Operator, go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.

2. Click **Add SIP Interface**.

3. Key in a name for the interface. The name must not contain spaces or special characters and must be unique.

4. Select **New provider**.

5. In the **With external number** field, key in the SIM card number and click **Next**.

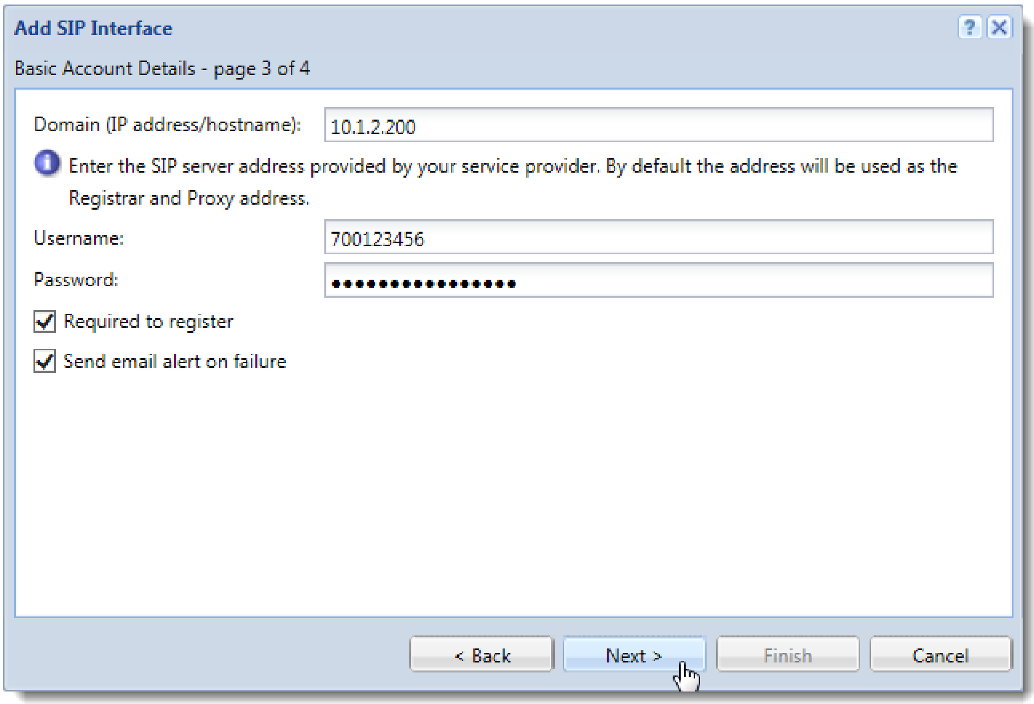
6. Select an extension that receives all calls.

7. In the **Prefix to dial out** field, key in a prefix for outgoing calls (7 in our example) and click **Next**.

8. In the **Domain (IP address/hostname)** field, key in the IP address of the NeoGate TG200 gateway.

9. In the **Username** field, key in the SIM card number and key in the **Password**configured for the VoIP account in the gateway.

10. Select the **Required to register** option and click **Next**.



11. Verify the information in the **Summary** section and click **Finish**.

You can now place some test calls. To make a call via the NeoGate TG200 gateway, use the prefix for outgoing calls configured for the gateway.

> **NOTE**
> Kerio Operator adds the interface prefix and extends the number (7+4411234567 in our example). To call back, you can dial the extended number because of the additionally configured outgoing route.

2.8.8 Configuring Kerio Operator and Yeastar NeoGate TE100 for calls over analog lines

Yeastar NeoGate TE100 is an analog-to-SIP gateway with a single E1/T1/J1 port that supports the PRI standard. You can use this device with Kerio Operator to make calls over the telephone network.

Learn how to integrate Kerio Operator with Yeastar NeoGate TE100.

Prerequisites

To complete the configuration, you need:

» Kerio Operator up and running.

» At least one internal extension.

» The Yeastar NeoGate gateway connected to the same LAN as Kerio Operator.

In the example below:

» The Kerio Operator IP address is `192.168.62.107`.

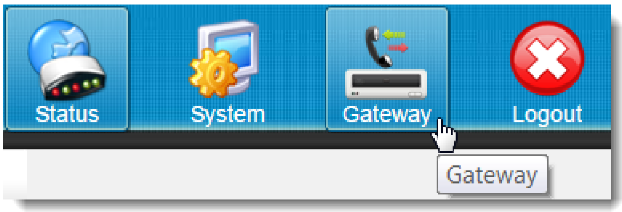
» The Yeastar NeoGate gateway's IP address is `10.1.2.200`.

» The trunk of numbers from the provider is `555 12xx`.

» The internal extension number is `100`.

Configuring Yeastar NeoGate TE100

1. In the administration interface of the Yeastar NeoGate gateway, click **Gateway**.



2. In the **Digital Trunk** section, click to open the trunk configuration.

3. In the **Mode Type** field, select:

   - `E1` in Europe.
   - `T1` in the USA.

3. In the **Linecoding** field, select:

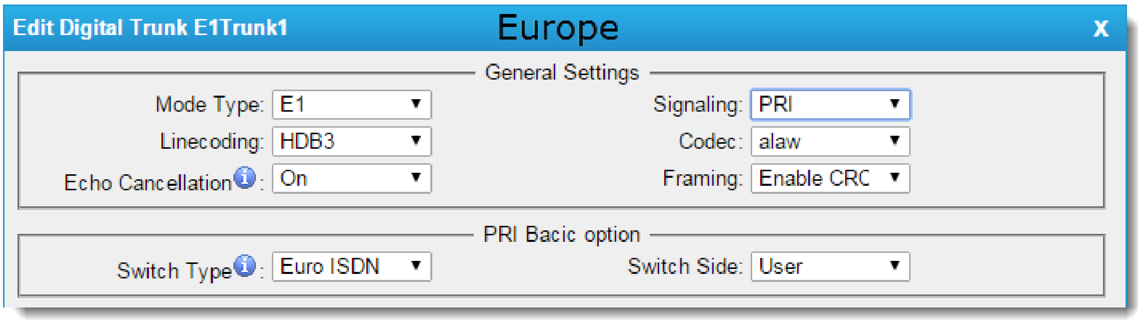
   - `HDB3` in Europe.
   - `B8ZS` in the USA.

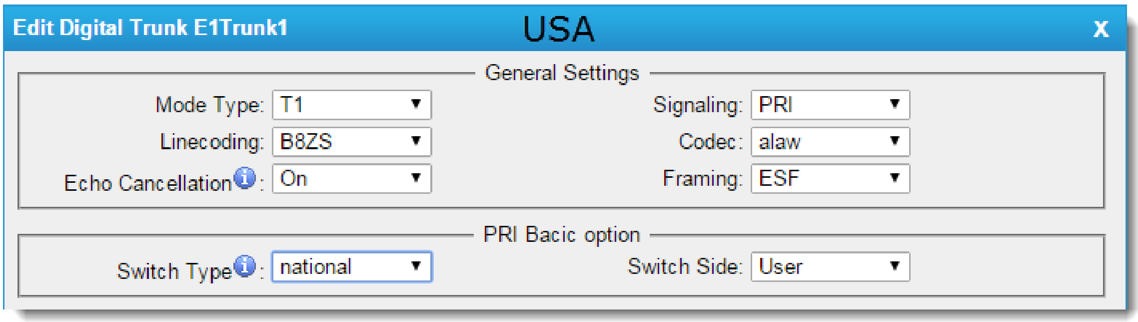
4. In the **Framing** field, select:

   - `Enable CRC4` in Europe.
   - `ESF` in the USA.

5. In the **Switch Type** field, select:

   - `Euro ISDN` in Europe.
   - `national` in the USA.





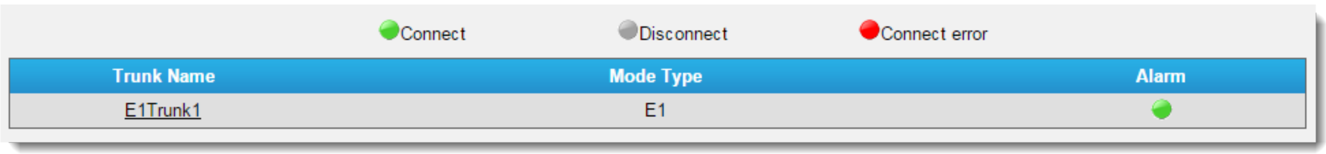
6. Click **Save**.

After you finish configuring the trunk, you need to verify that the gateway is connected:

1. Click the **Status** button in the upper right corner.

2. Go to **System Status > E1/T1 Status**.

3. Verify that **Alarm** indicates **Connect**.



After verifying the gateway connection, continue with the configuration:

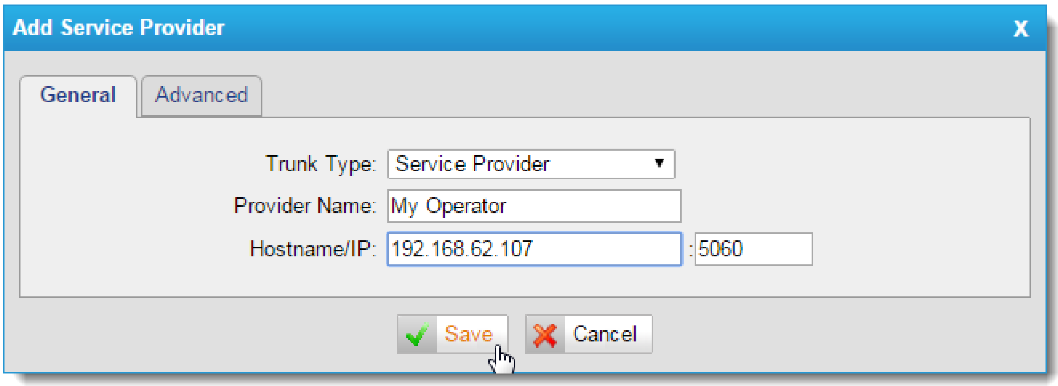
1. Go to **VoIP Settings > VoIP Trunk** and remove all default VoIP trunks.

2. Click **Add VoIP Trunk** to open the **Add New Account** dialog box.

3. In the **Trunk Type** field, select `Service Provider`.

4. Key in a name for the VoIP trunk. For example, `My Operator`.

5. Key in the IP address (`192.168.62.107`) of Kerio Operator.



6. Click **Save**.

7. Go to **Route Settings > Route List**.

8. Click to edit the `E1_to_SIP` route.

9. In the **Send Call Through** field, select `ServiceProvider — MyOperator` and click **Save**.

10. Click to edit the `SIP_to_E1` route.

11. In the **Call Comes in From** field, select `ServiceProvider — MyOperator` and click **Save**.

Route name	Simple Mode	Call Comes in From	Send Call Through		
E1_to_SIP	yes	E1Trunk1	MyOperator		
SIP_to_E1	yes	MyOperator	E1Trunk1		

12. Click **Apply Changes** in the upper right corner.

Configuring Kerio Operator

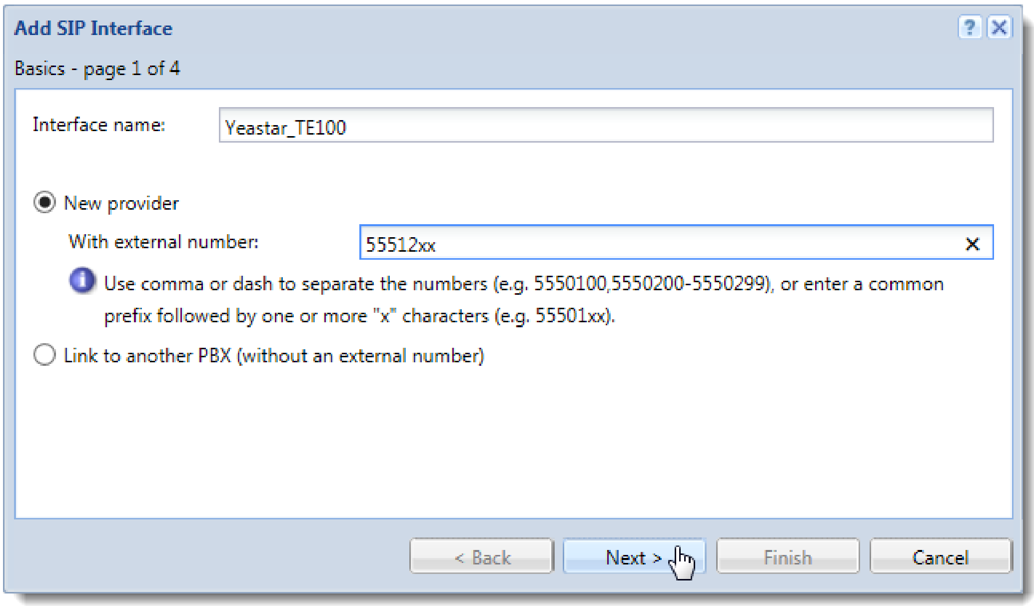
1. In the administration interface of Kerio Operator, go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.

2. Click **Add SIP Interface**.

3. Key in a name for the interface (for example, the name of the gateway). The name must not contain spaces or special characters and must be unique.

4. Select **New provider**.

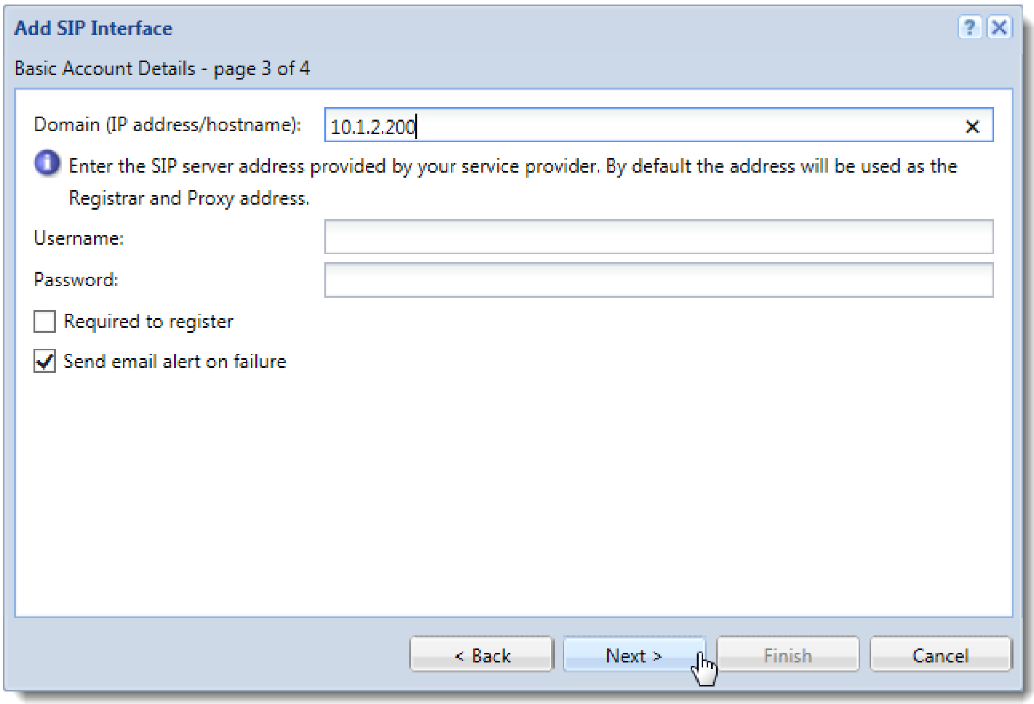
5. In the **With external number** field, key in the number `555 12xx` and click **Next**.



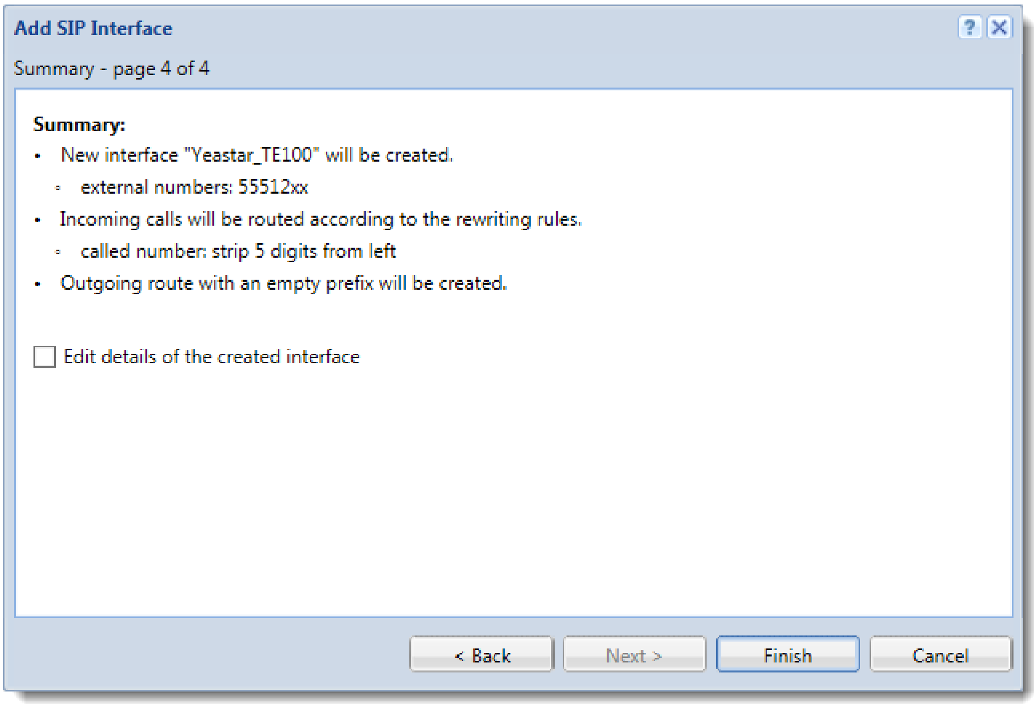
6. Select an extension (`100`) that receives all calls.

7. Optionally, in the **Prefix to dial out** field, key in a prefix for outgoing calls and click **Next**.

8. In the **Domain (IP address/hostname)** field, key in the IP address (`10.1.2.200`) of the Yeastar NeoGate gateway.



9. Verify the information in the **Summary** section and click **Finish**.



2.8.9 Configuring PRI telephone service through the Digium VoIP Media Gateway

You can connect Kerio Operator to a Primary Rate Interface (PRI) telephone service. This requires a gateway device or adapter to convert the voice media into a signal that Kerio Operator can process. The following topic covers the necessary steps to configure Kerio Operator with the Digium G200 VoIP Media Gateway.

Installing the Digium VoIP Media Gateway on the network

The device obtains an IP address automatically when connecting to the network. You can identify the device's IP address from your network DHCP server. If you use Kerio Operator as the DHCP server you can locate the leased IP address from **Configuration > Network > [DHCP leases button].**

Connecting to the Digium VoIP Media Gateway

1. Input the IP address of the device into the web browser of a management computer located on the same network. You must connect using a secure (HTTPS) type of connection.

2. Login to the device as **Admin** with the password **Admin**

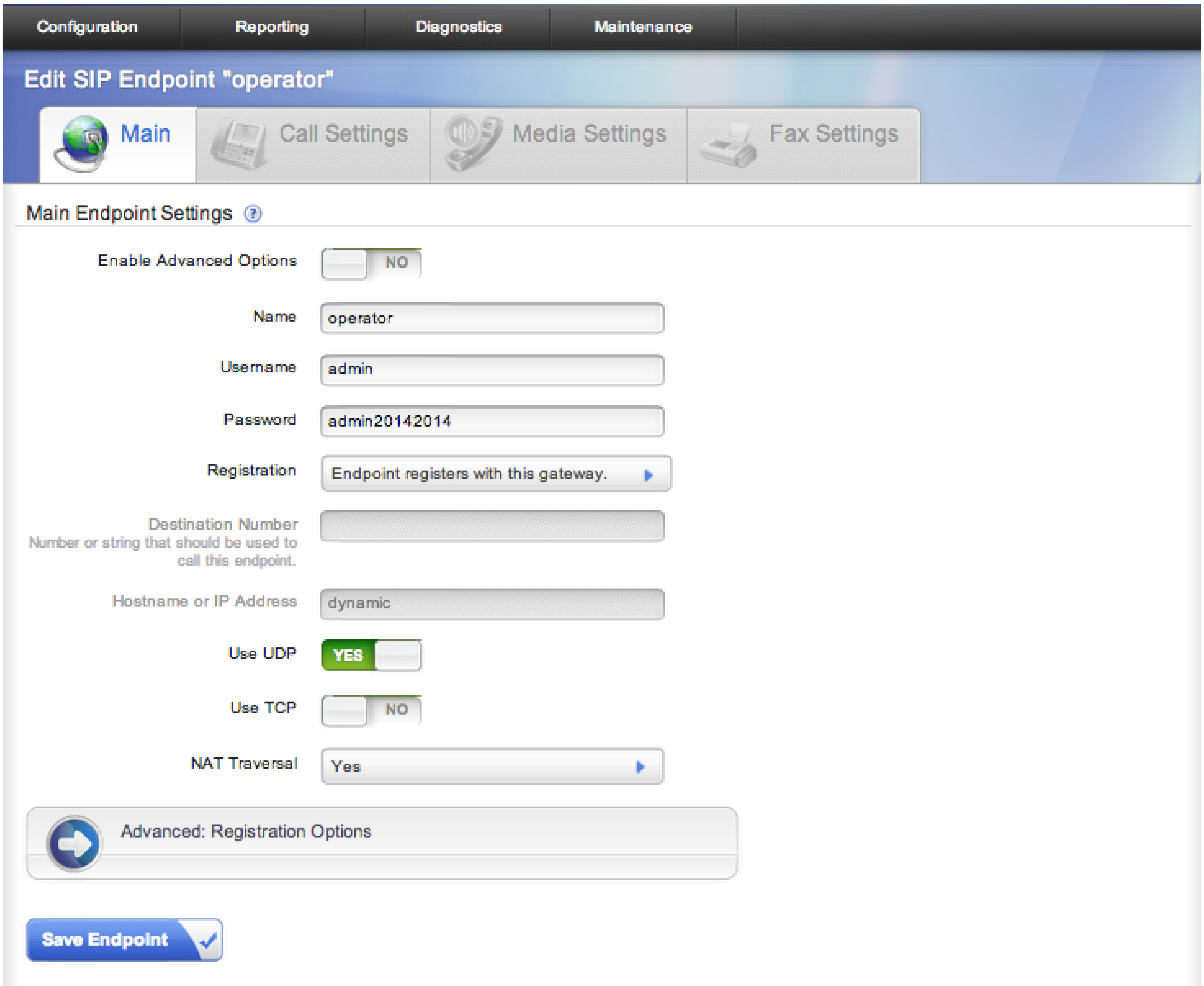
3. After logging into the device, you can assign a static IP address and configure a new administration password. Refer to the Gateway User's Manual for details.

Configuring the Digium VoIP Media Gateway

Adding a SIP endpoint

A SIP endpoint defines the credentials that Kerio Operator uses to register with the gateway device.

1. Locate **Configuration > SIP endpoints**
2. Add a new SIP endpoint and input the following parameters:
   - *Name*: Custom name for the SIP endpoint (e.g., **operator**).
   - *Username*: The SIP username that Kerio Operator uses to register.
   - *Password*: The SIP password that Kerio Operator uses to register.
   - *Registration*: Select **Endpoint registers with this gateway**.
   - All other parameters should use the default setting.



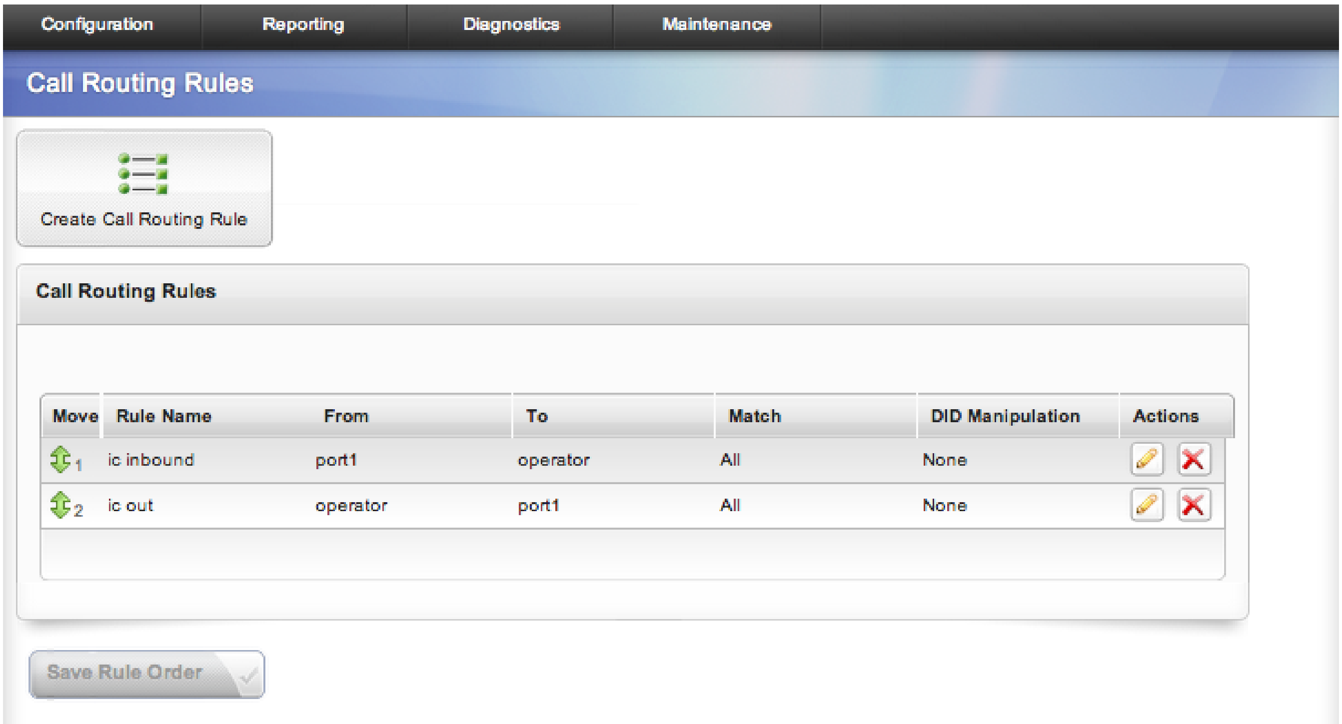
Adding call routing rules

The Digium device can support multiple SIP endpoints and PRI/BRI interfaces. Therefore it is necessary to create rules that define where to route incoming and outgoing calls. Locate **Configuration > Call Routing Rules**.

Create a rule to direct calls from the PRI/BRI interface to the SIP endpoint:

- *From*: Choose the PRI/BRI port (e.g., **port1**).
- *To*: Choose the SIP endpoint (e.g., **operator**).
- *Match*: Choose **All**.
- *DID* Manipulation: **None**.

Create another rule to direct calls from the PRI/BRI interface to the SIP endpoint. Use the same parameters as the previous rule, with the **From** and **To** values in reverse order.



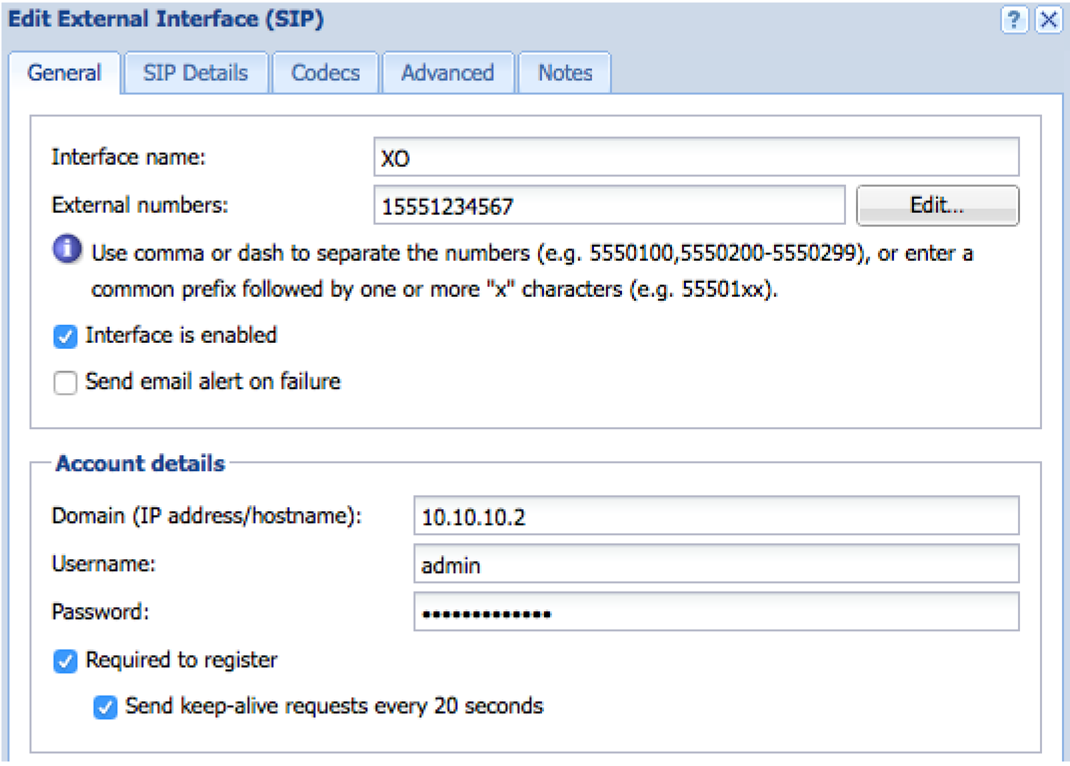
> **NOTE**
> If you have multiple PRI/BRI interfaces you should create a Call Routing Group that includes each connected PRI interface. Refer to this group when configuring Call Routing Rules.

Registering Kerio Operator to the Digium device

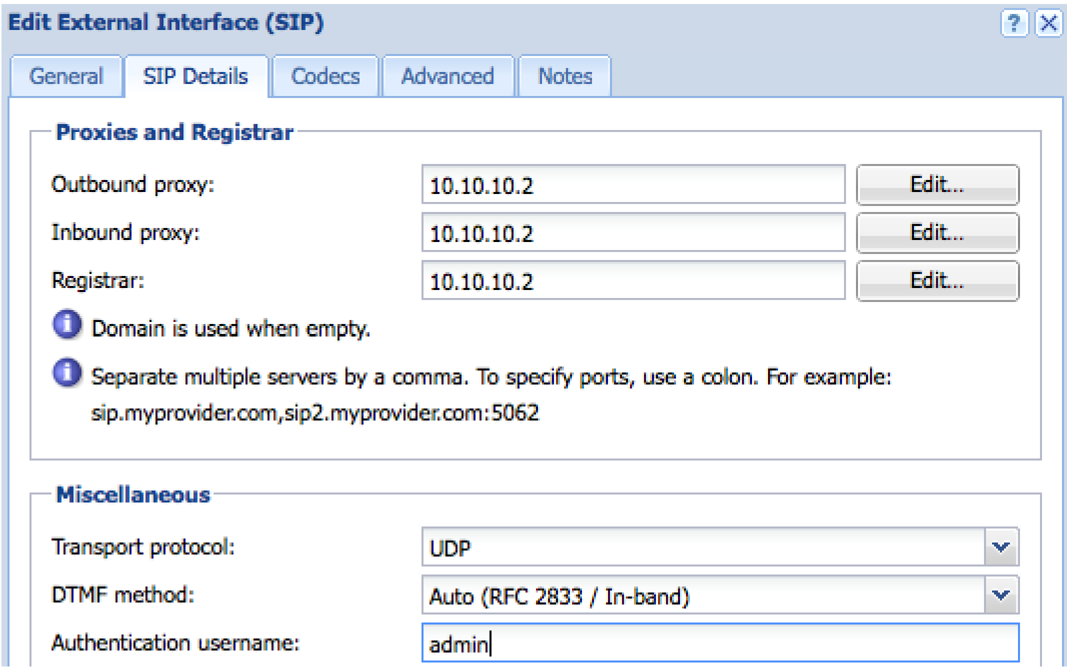
Locate **Configuration > Call Routing** in the Kerio Operator administration. Add a SIP interface and configure the following parameters:

- *Interface name*: A label for the SIP interface.
- *New provider with external number*: Input the phone number(s) assigned by your telephone service provider.
- *Incoming calls*: Specify where to route incoming calls.
- *Outgoing calls*: Define a prefix if desired.
- *Domain, hostname or IP address*: IP address of the Digium device.
- *Username*: The username assigned to the SIP endpoint.

- *Password*: The password assigned to the SIP endpoint.
- *Required to register with Registrar*: Enable this option.



After you add the SIP interface, edit the interface and go to the **SIP details** tab. Locate the **Authentication** username field and define the username of the SIP endpoint.

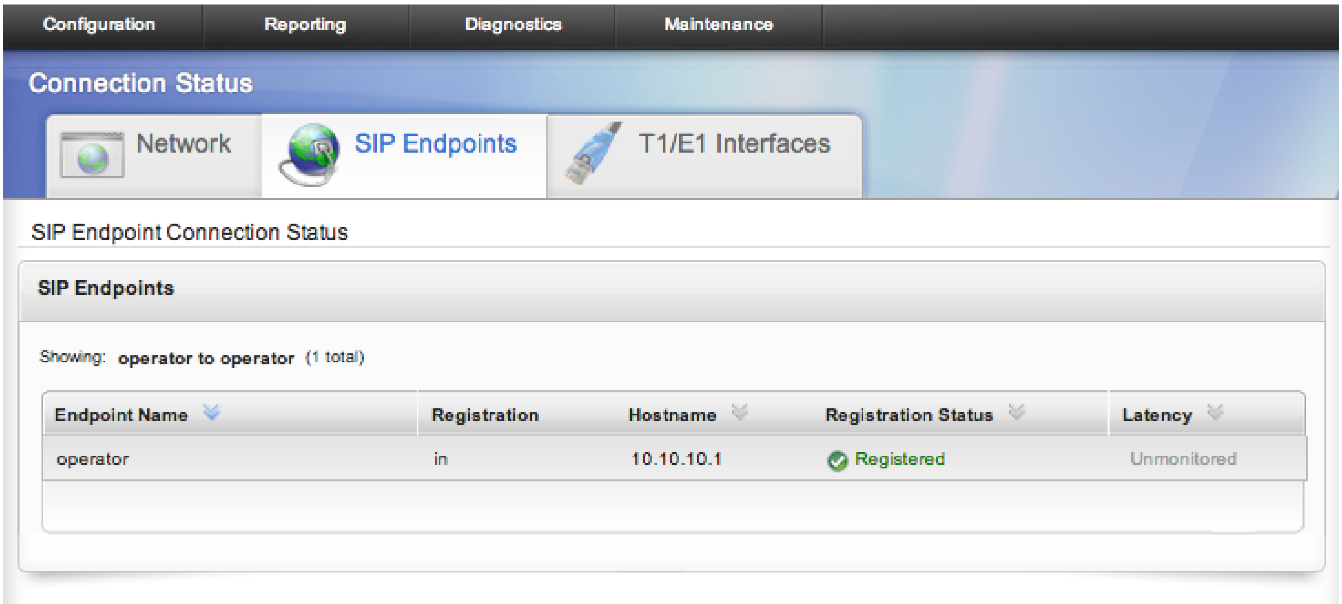


NOTE

You may need to contact your telephone service provider to ensure that they do not remove digits from the telephone number when routing incoming calls to your PRI service. You can refer to the security log to verify the details of rejected calls.

Checking connectivity status

You can review the status of the SIP endpoints and PRI ports from the administration of the Digium device. Locate the **Diagnostics** dialog. Review the **Network, SIP Endpoints**, and **T1/E1** Interfaces to verify successful connectivity of all configured items.



2.8.10 Connection with Linksys SPA3102 analog (FXS/FXO) to SIP gateway

Linksys SPA3102 is an analog-to-SIP gateway equipped with one FXS and one FXO telephone port.

> **NOTE**
> This information is designed for Kerio Operator 2.4 and newer.

Learn how to configure SPA3102 gateway with Kerio Operator over the FXO interface.

Prerequisites

To complete the configuration, you need:

- Kerio Operator up and running.
- At least one internal extension.
- The PSTN line connected to the **Line** port of the SPA3102 device.
- The SPA3102 gateway connected to your LAN through only one port.

IMPORTANT

The Linksys SPA3102 gateway has its own DHCP server in the **Ethernet** (LAN) interface and a DHCP client in the **Internet** (WAN) interface. If you already have another DHCP server in your LAN, do not connect the Ethernet interface of the gateway. Follow the steps in the installation guide of the device.

After the installation of the device:

1. Connect the **Internet** port to your LAN.

2. Connect an analog phone to the **Phone** port of the SPA3102 gateway.

3. Dial `****` to access the voice menu.

4. Dial `110#` to obtain the IP address of the device.

5. To enable web administration on the WAN port of the SPA3102 gateway, dial `7932# 1#`.

In the example below:

- The Kerio Operator IP address is `10.1.2.95`.
- The Linksys SPA3102 gateway IP address is `10.1.2.200`.
- The PSTN number is `123456`.

Configuring Kerio Operator

1. In the Kerio Operator administration interface, go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.

2. Click **Add SIP Interface**.

3. Key in a name for the interface. The name must not contain spaces or special characters and must be unique.

4. Select **New provider**.

5. In the **With external number** field, key in the PSTN number and click **Next**.

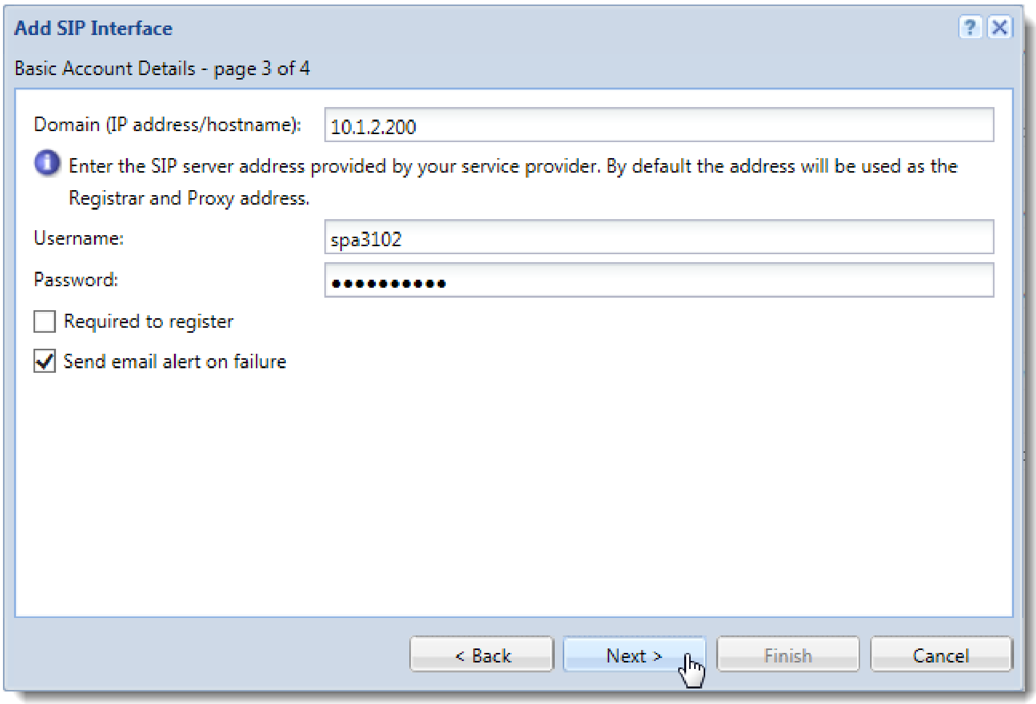
6. Select an extension that receives all calls.

7. In the **Prefix to dial out** field, key in a prefix for outgoing calls (9) and click **Next**.

8. In the **Domain (IP address/hostname)** field, key in the IP address of the Linksys SPA3102 gateway.

9. Key in an username (`spa3102`) and a password (`pass123456`).

10. Disable the **Required to register option** and click **Next**. The gateway does not behave as a SIP registrar, so Kerio Operator must authenticate before each call.



11. Select the **Edit details of the created interface** option and click **Finish**.

After you finish the configuration, the **Edit External Interface** dialog box opens:

1. Go to the **SIP Details** tab.

2. In the **Outbound proxy** field, key in the IP address of the gateway and the port, `10.1.2.200:5061`. The **Inbound proxy** field leave empty.

3. Click **OK** to save your settings.

Configuring the Linksys SPA3102 gateway

1. Connect to the Linksys SPA3102 web administration interface and go to **Voice > Advanced**:

2. Go to **Line 1**, in the **Line Enable** field select **no**, and click **Submit All Changes** to save your settings.

3. Go to **PSTN Line** and in the **Line Enable** field select **yes**.

4. In the **SIP Port** field, key in `5061`.

5. Go to the **Proxy and Registration** section:

   a. In the **Proxy** field, key in the Kerio Operator IP address.

   b. In the **Use Outbound Proxy** field, select **no**.

   c. In the **Register** field, select **no**.

   d. In the **Use OB Proxy In Dialog** field, select **no**.

   e. In the **Make Call Without Reg** field, select **yes**.

   f. In the **Ans Call Without Reg** field, select **yes**.

5. Go to the **Subscriber Information** section:

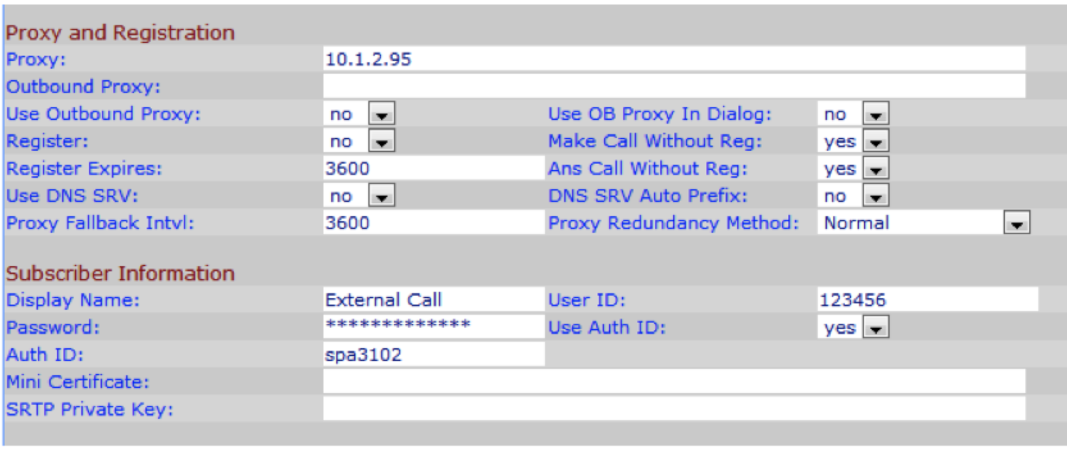
a. In the **Display Name** field, key in a new name (`External Call` in our example).

b. In the **User ID** field, key in the PSTN number.

c. In the **Password** field, key in the password configured for the SIP interface in Kerio Operator.

d. In the **Use Auth ID** field, select **yes**.

e. In the **Auth ID** field, key in the username configured for the SIP interface in Kerio Operator.



6. Go to the **Dial Plans** section and in the **Dial Plan 8** field key in the `S0<:123456@10.1.2.95>` string.

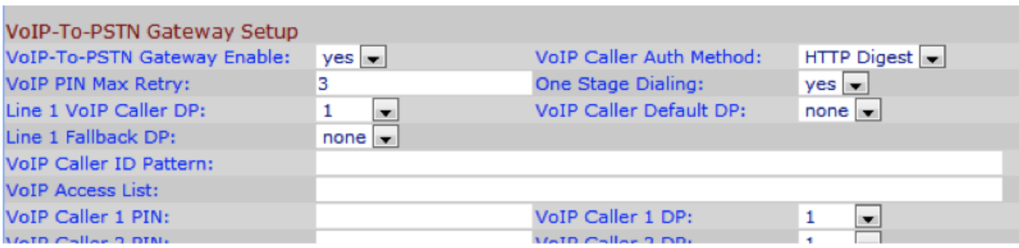
7. Go to the **VoIP-To-PSTN Gateway Setup** section:

a. In the **VoIP-To-PSTN Gateway Enable** field, select **yes**.

b. In the **VoIP Caller Auth Method** field, select **HTTP Digest**.

c. In the **One Stage Dialing** field, select **yes**.

d. In the **VoIP Caller Default DP** field, select **none**.



8. Go to the **VoIP Users and Passwords (HTTP Authentication)** section:

a. In the **VoIP User 1 Auth ID** field, key in the username configured for the SIP interface in Kerio Operator.

b. In the **VoIP User 1 Password** field, key in the password configured for the SIP interface in Kerio Operator.

c. In the **VoIP User 1 DP** field, select **none**.

9. Go to the **PSTN-To-VoIP Gateway Setup** section:

a. In the **PSTN-To-VoIP Gateway Enable** field, select **yes**.

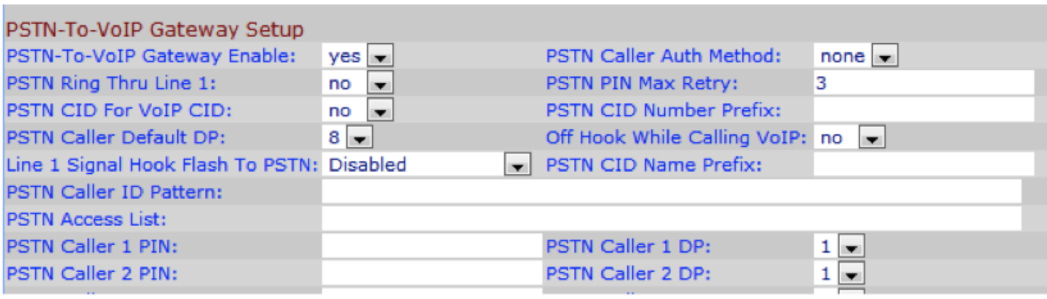
b. In the **PSTN Caller Auth Method** field, select **none**.

c. In the **PSTN Ring Thru Line 1** field, select **no**.

d. Leave the **PSTN CID Number Prefix** option blank.

e. In the **PSTN Caller Default DP** field, select **8**.

> **NOTE**
> If your PSTN line provides the Caller ID service, in the **PSTN CID For VoIP CID** field, select **yes**, and increase the value for **PSTN Answer Delay** (see step below), so the SPA3102 gateway can transmit the Caller ID using the FSK modulation between the first and the second ring of the call before the start of the VoIP call to Kerio Operator.



10. Go to the **FXO Timer Values (sec)** section:

    a. In the **VoIP Answer Delay** field, key in `0`.

    b. In the **PSTN Answer Delay** field, key in `0`.

> **NOTE**
> To transfer the PSTN Caller ID to the VoIP side, key in the number of seconds that represents two rings on your PSTN line.

11. Click **Submit All Changes** to save your settings.

2.9 Kerio Operator API

The Kerio Operator API enables you to programmatically access your Kerio Operator server to integrate with third-party solutions or write scripts to automate specific tasks. The API provides all actions available in the client and administration interfaces of the product. For example, you can add/remove users, update IP address groups, read logs, manage time ranges, and much more.

Licensing

The API is part of Kerio Operator and is governed by the product's license. In summary, the license to use Kerio Operator also entitles you to use the API on your server. Third-party software vendors may distribute their applications both for free or for profit. There is no royalty fee for using the API.

Get involved

A dedicated forum is available for sharing your ideas, questions, answers, and other feedback with the developer community. You can participate in the forum at forums.kerio.com.

Introduction

The API is accessible via a secure, session based HTTP connection. All communication is formatted as a JSON string and directed to a designated URL. For example, a request for the Kerio Operator administration API may go to:

```
https://server.example.com:4021/admin/api/jsonrpc/
```

The URL to access the API consists of the following parts:

URL part	Title	Description
`https://`	Protocol	The API uses HTTPS
`server.example.com`	Hostname	The Internet accessible hostname (domain name) of your server
`:4021`	Port	The port number for accessing the API
`/admin/api/jsonrpc/`	File path	The file path of the JSON interface

The Kerio Operator API is accessible via two separate interfaces that provide different functionality. The Kerio Phone interface provides phone functionality such as adding favorites, reviewing recent calls, or setting up call forwarding. The Kerio Operator Administration interface configures all server based functionality such as adding users, configuring SIP interfaces, or assigning extensions. The port number and file path vary depending on the interface.

Refer to the table below:

Product interface	Port	File path
Kerio Phone	443	/myphone/api/jsonrpc/
Kerio Operator Administration	4021	/admin/api/jsonrpc/

Prerequisites

To access the API, you need a Kerio Operator user account. Depending on the interface, this may be a standard user account, or an administrator account with elevated permissions. The type of account you use to authenticate determines your permissions when accessing the API. In order to communicate with the API you need a framework for processing the requests and managing the network connection. A common framework for these actions is PHP and cURL.

Authentication

The API uses a two step authentication process. The first authentication step involves a username and password, the same type of account used to access the Kerio Operator interface. After successful authentication, the API returns a randomized token, and a session cookie. All subsequent requests to the API must include the token and session cookie in the HTTP headers of the connection. The method for handling the session cookie and token depends on the framework used in your application. Refer to the example below for information on how you can manage authentication to the API using cURL.

API Requests

All requests to the API must be encoded as a JSON string. The JSON string is an array consisting of the JSON version, an identifier, the API method, and the parameters of the query. You can use a web browser to inspect the JSON request and response for any interaction with the product interface. Refer to Inspecting the Kerio Operator API communication in a web browser for details.

Example API Communication

The following example uses PHP and cURL to get the next available extension number.

```
//Custom variables
$api_cookie="/tmp/kerio-api-cookie";
$api_url="https://operator.example.com:4021/admin/api/jsonrpc/";
$api_user="admin-api";
$api_password="secretpassword”;
```

```
//Initialize the cURL request
$ch = curl_init();
curl_setopt($ch, CURLOPT_URL, $api_url);
curl_setopt($ch, CURLOPT_HTTPHEADER, "Content-Type:ap-
plication/json");
curl_setopt($ch, CURLOPT_RETURNTRANSFER, true);
curl_setopt($ch, CURLOPT_SSL_VERIFYHOST, false);
curl_setopt($ch, CURLOPT_SSL_VERIFYPEER, false);
curl_setopt($ch, CURLOPT_COOKIEJAR, $api_cookie);
curl_setopt($ch, CURLOPT_COOKIEFILE, $api_cookie);

//Server login request to obtain the API token and session cookie
$api_login = array(
  'jsonrpc' => '2.0',
  'id' => 1,
  'method' => 'Session.login',
  'params' => array(
    'userName' => $api_user,
    'password' => $api_password,
    'application' => array(
      'name' => 'Sample app',
      'vendor' => 'Kerio',
      'version' => '1.0'
    )
  )
);
curl_setopt($ch, CURLOPT_POSTFIELDS, json_encode($api_login));
$return=json_decode(curl_exec($ch),true);
$token=$return['result']['token'];

//Verify the token, otherwise return the JSON response and exit
the script.
if ($token=="") {printf(htmlspecialchars($json_return)); exit();}

//Add the token to the cURL headers
curl_setopt($ch, CURLOPT_HTTPHEADER, array("Content-Type:ap-
plication/json","X-Token:".$token));

//Get the next available extension number
$api_query = array(
  'jsonrpc' => '2.0',
  'id' => '1',
  'method' => 'Extensions.getNextValidNumber',
  'params' => array()
);
curl_setopt($ch, CURLOPT_POSTFIELDS,json_encode($api_query));
$return=json_decode(curl_exec($ch),true);
```

```
$next_ext=$return['result']['extensionNumber'];

curl_close($ch);
```

2.9.1 Inspecting Kerio Operator API communication in a web browser

You can use a web browser to inspect the API communication used in Kerio Operator. This is helpful in case you want to learn how to adapt functionality into your custom application.

The web interfaces of Kerio Operator use XMLHttpRequest (XHR) to exchange JSON formatted data. Browsers display this type of activity in the network details section of the developer view. You can find each JSON formatted request and the corresponding response of any action you perform in the interface.

1. Log in to the Kerio Phone or Kerio Operator administration interface.

2. Enable the developer view in your browser.

3. Perform the action you want to build into your application.

4. Locate the JSON requests and responses in your browser's developer view.

Using Google Chrome to obtain the API request for adding a user in Kerio Operator

1. In Google Chrome, enable the Developer Tools.

2. Log in to the Kerio Operator administration and create a new user. For more information, refer to Creating user accounts (page 192).

3. In the Google Chrome developer view, go to the **Network** panel. See Network panel overview for details.

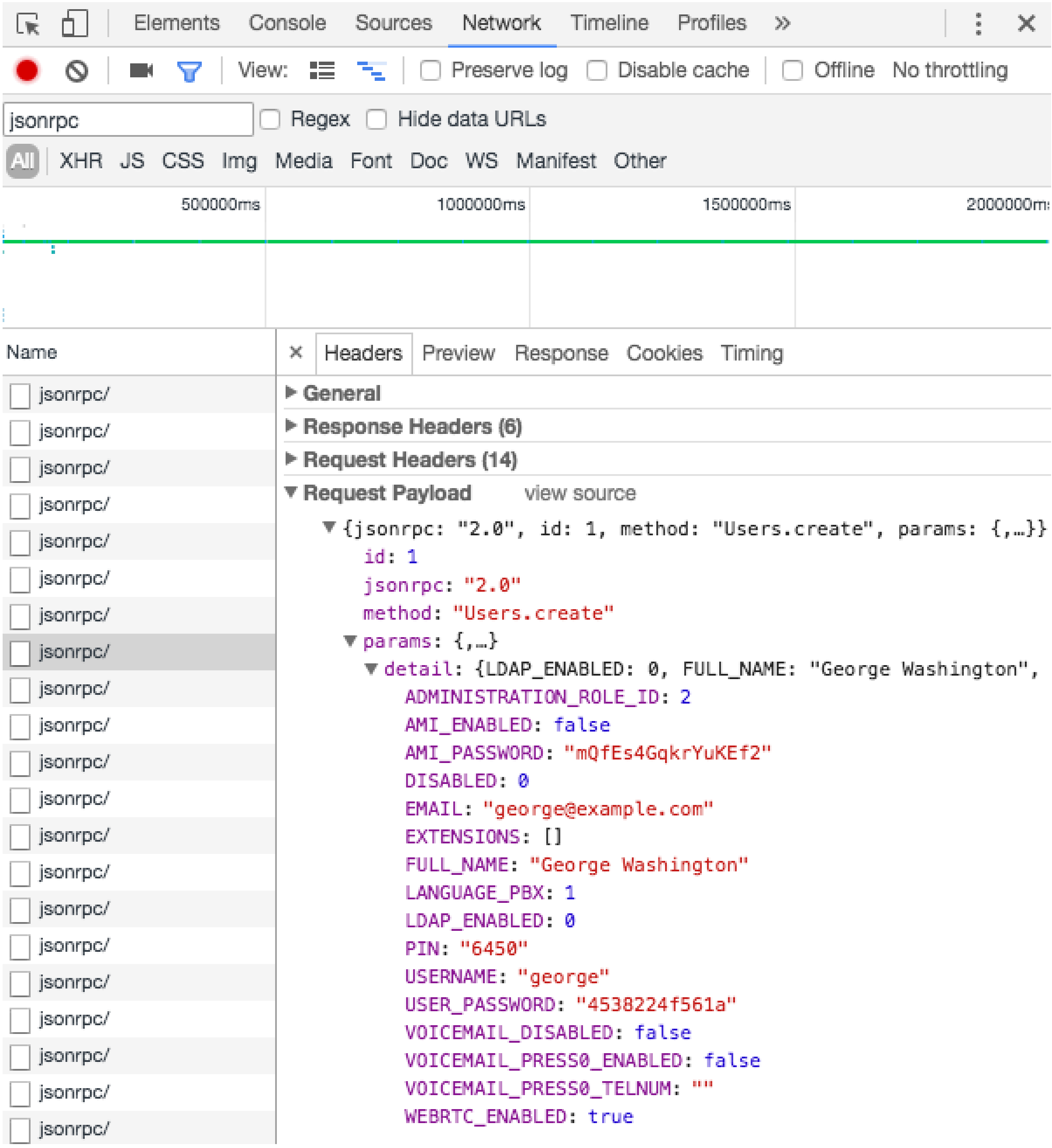
4. Locate the resources named `jsonrpc/` in the **Requests Table** and select the first item. Note that you can also filter the table results to display only API resources. In the filter box, type the string "jsonrpc".

5. In the preview area of the resource, select the **Headers** tab.

6. Expand **Request Payload** to view the formatted JSON request.

7. Browse each JSON resource and examine the payload details until you locate the resource that corresponds to your action.

8. Click **view source** to show the unformatted JSON string.



Screenshot 16: JSON resources showing in Developers Tools

3 Using

This section contains information about:

3.1 Hardware phones and devices

This section helps you configure and use hardware phones.

3.1.1 Hardware telephone basic usage

Learn how to use common functions of hardware phones supported by Kerio Technologies).

Using loud speaker

Every telephone has a special button for loud speaker (speaker button). You can usually press this button either before dialing or anytime during a call.

Using Do not Disturb

If you select Do not Disturb option (DnD), the phone will generate the busy tone to inform the other party that you are not available.

Learn how to configure DnD in various supported hardware phones.

Cisco IP Phone 7960/7940

1. Press the **Settings** button

.

2. Scroll to **Call** Preferences and press **Select**.

3. Scroll to **Do Not Disturb**.

4. Press **Yes**.

Cisco SPA

1. Press **more**.

2. Press **DnD**.

Polycom IP 33x

1. Press **Menu**.

2. Select **Features**.

3. Select **Do not disturb**.

For unblocking incoming calls, use the same sequence.

Forwarding calls

Most hardware phones support call forwarding. However, you can also set it in Kerio Phone too.

Conference calls

Most hardware phones also support conference calls. However, it is much easy and simple to use conferences in Kerio Operator.

Transferring calls

When you want to transfer a call, you usually have two options:

» Attended transfer -you can connect with the third party, find out if the person is on the phone and make an announcement.

» Blind transfer - the call is transferred without any announcement.

Learn how to configure these options in various supported hardware phones.

Cisco IP Phone 7960/7940

Attended transfer

1. Initiate or answer the call.

2. Press **more**.

3. Press **Transfer**.

4. Dial the number to which you want to transfer the call.

5. When the dialed number rings, press **Transfer** again, or wait for the other party's answer, announce the call and then press **Transfer**.

6. Hang up if the party accepts the call.

If the party refuses the call, return to the original call by pressing the **Resume** softkey.

Blind transfer

1. Initiate or answer the call.
2. Press **more**.
3. Press **BlndXfr**.
4. Dial the number to which you want to transfer the call.
5. Terminate the call.

Cisco SPA

Attended transfer

1. Initiate or answer the call.
2. Press **Trnsfer**.
3. Dial the number to which you want to transfer the call.
4. Wait for the connection.
5. Press **Trnsfer** again.
6. Hang up.

Blind transfer

1. Initiate or answer the call.
2. Press **Trnsfer**.
3. Dial the number to which you want to transfer the call.
4. Press **Trnsfer** again.
5. Hang up.

Polycom IP 33x

Attended transfer

1. Initiate or answer the call.
2. Press **Trans**.
3. Dial the number to which you want to transfer the call.
4. Wait for the connection.
5. Press **Trans** again.
6. Hang up.

Blind transfer

1. Initiate or answer the call.
2. Press **Trans**.

3. Press **Blind**.

4. Dial the number to which you want to transfer the call.

5. Hang up.

Snom 300

Attended transfer

1. Initiate or answer the call.

2. Press **L1/L2** key (hold).

3. Dial the number to which you want to transfer the call.

4. Wait for the connection.

5. Press **Transfer** key.

6. Press **OK** key.

7. Hang up.

Blind transfer

1. Initiate or answer the call.

2. Press **Transfer** key.

3. Dial the number to which you want to transfer the call.

4. Press **OK** key.

5. Hang up.

Snom 320/821

Attended transfer

1. Initiate or answer the call.

2. Press **Hold** key.

3. Dial the number to which you want to transfer the call.

4. Wait for the connection.

5. Press **Transfer** key.

6. Press **OK** key.

7. Hang up.

Blind transfer

1. Initiate or answer the call.

2. Press **Transfer** key.

3. Dial the number to which you want to transfer the call.

4. Press **OK** key.

5. Hang up.

3.1.2 Configuring BLF on Polycom phones

BLF (Busy Lamp Field) is a light placed on a telephone device. BLF notifies you if another extension is busy or not.

Setting BLF on Polycom telephones is not as easy as on other telephones. You need to prepare a configuration file and upload it to Kerio Operator.

Preparing a configuration file

Prepare a file with the following content:

```
<?xml version="1.0" encoding="UTF-8" standalone="yes"?><attendant
attendant.resourceList.1.address="sip:10@192.168.12.91"
attendant.resourceList.1.label="Ten"
attendant.resourceList.2.address="sip:11@192.168.12.91"
attendant.resourceList.2.label="Eleven"/> <call
call.directedCallPickupMethod="legacy" call.directedCallPickupString="*8"/>
```

This example file enables two BLF buttons. Button one monitors extension `10` on Kerio Operator with IP address `192.168.12.91`. This button is labeled `Ten`. A second button monitors extension `11` on same Kerio Operator and has label `Eleven`. Call pickup PBX service is set to `*8`.

Upload the file to `/var/tftp/polycom-0004f223510b-manual.cfg` where `0004f223510b` is a phone hardware (MAC) address.

Uploading the configuration file to Kerio Operator

> **NOTE**
> The file must be uploaded via SSH using SCP.

Enabling SSH in Kerio Operator

Follow these instructions:

1. In the administration interface, go to **Status > System Health**.

2. Click **Tasks** while pressing the **Shift** key.

3. Select **Enable SSH**.

4. Connect to Kerio Operator via SCP (use for example WinSCP for Windows). For access, use username `root` and password of a Kerio Operator administrator.

5. Upload the file.

3.1.3 Configuring Cisco / Linksys SPA phones to support more than three callers in a conference

The Cisco / Linksys SPA phone models support 3-way calling, which allows up to three callers in a conference. In some cases, you may require more than three callers. In this scenario, administrators can configure conferences. To further simplify the workflow, you can adjust the conference button of Cisco / Linksys phones to utilize a pre-configured conference line. This allows the operator of the Cisco / Linksys phone to move callers into the conference, without requiring the caller to hang up and call back into the conference extension.

Configuring dedicated conference extensions

Each person who may need to incorporate this type of conferencing should be assigned their own static conference extension. For example, Alice may have extension 220, and Bob has extension 221. Their dedicated conference extensions may be 3220 (for Alice) and 3221 (for Bob).

Adjusting the phone's conference button

Access the phone's web administration by entering its IP address into a browser (from a computer on the same network). You can obtain the phone's IP address through the soft menu of the device, or from the 'Extensions' dialog in the Kerio Operator administration. After entering the phone's administration, follow these steps:

1. Click the **Admin login** link in the top right corner (If the phone is automatically provisioned, you may be prompted to login).

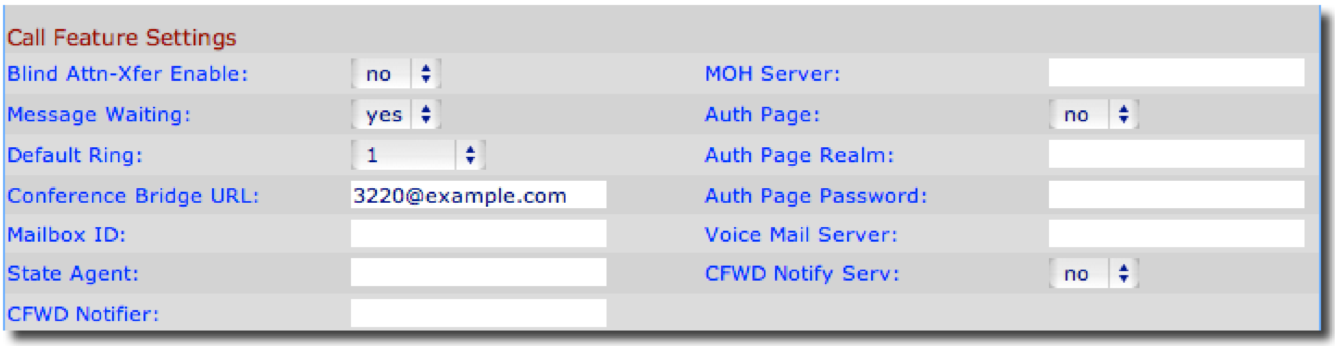
2. If you are required to login, the user is 'admin' and the password is the master password assigned in the **Configuration > Provisioned Phones** dialog (Additional information is available in Configuring automatic phone provisioning).

3. Click the **advanced** link.

4. Go to the **Ext 1** tab.

5. Locate the **Conference Bridge URL** input field.

The value of the **Conference Bridge URL** is the extension of the conference at (@) the host / IP of Kerio Operator. For example, Alice's dedicated conference is 3220, so her conference bridge URL would be 3220@example.com (where example.com is the hostname of your Kerio Operator). After inputting the value, click **Submit All Changes** at the bottom of the page and allow the phone to reboot.



Operating conferences from the phone

By specifying a conference bridge URL, the action of the conference soft key will change, however the functional behavior remains the same. To place callers into the conference, follow these steps:

1. Press the **conf** soft key while on an active call (this will place the caller on hold)

2. Make a new outgoing call

3. Once connected to the new party, press the 'confLx' soft key. This will move all parties to the conference.

4. Repeat steps 1 - 3 for each additional caller.

3.1.4 Configuring Snom M300/M700 with Kerio Operator

Snom M300/M700 is a SIP-to-DECT base station working with M-series wireless handsets.

NOTE
This information is designed for Kerio Operator 2.4 and newer.

Kerio Operator provisions Snom M300 and M700 base stations, but you must register handsets and assign internal extensions to them to make and receive calls.

Prerequisites

- The base station connected to Kerio Operator.
- At least one Snom handset.
- At least one internal extension assigned to the phone.

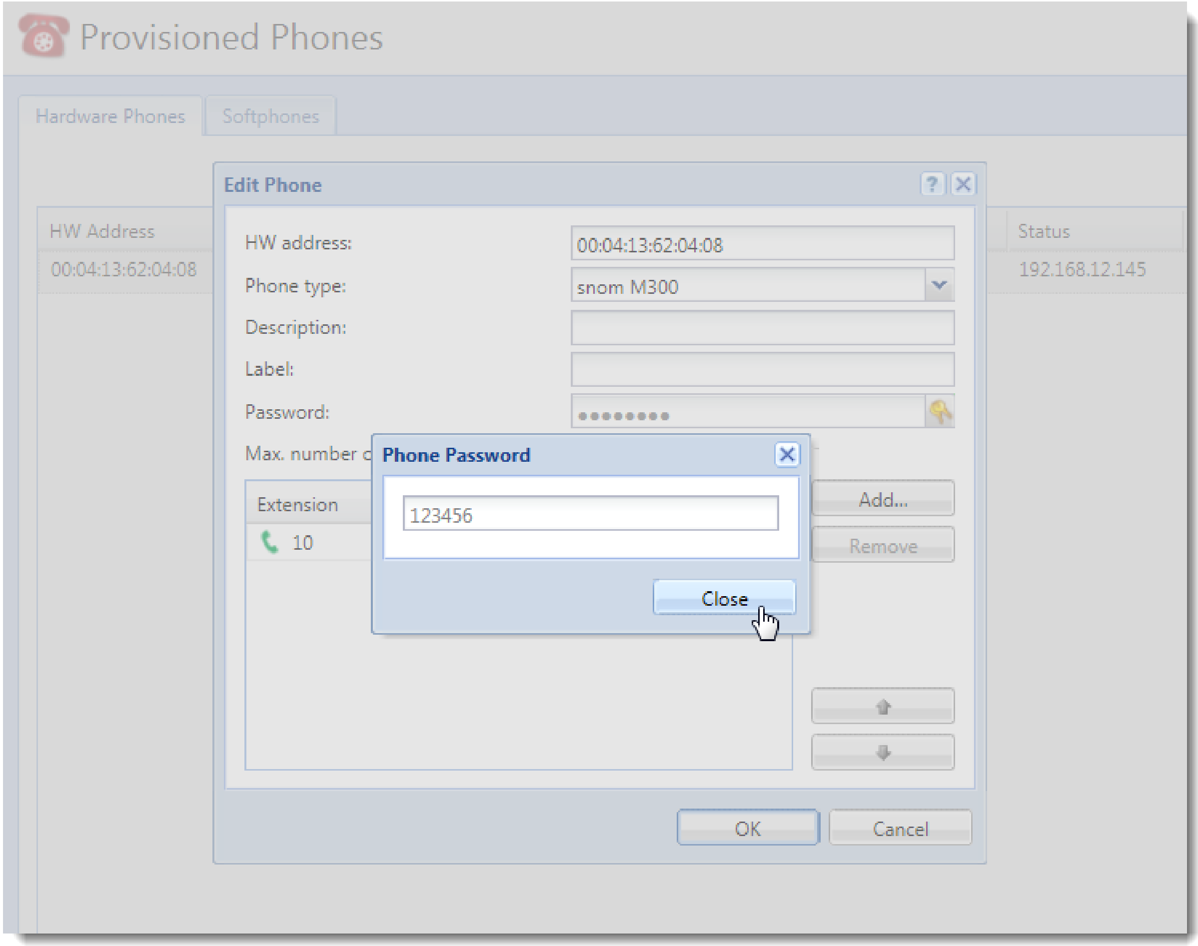
To assign an extension to the phone:

1. Go to **Provisioned Phones**.
2. Select a phone and click **Edit**.
3. Click **Add** and assign an extension.
4. Click **OK** to save your changes.

For more information, refer to How to add a phone (page 171).

Configuring the base station and the handset

After Kerio Operator provisions the base station and you receive the IP address and password, you need to register the handsets and then assign the internal extensions in the base station's administration interface.



Registering the handset

To see the handset in the base station's administration interface, you must register the handset first:

1. Open the main menu of the Snom handset and go to **Connectivity**.

2. Select **Register**.

3. Key in a password of the handset and click **OK**.

The handset then registers to the base station.

Assigning the internal extension to the handset

After you register the handset to the base station, you need to assign an extension to the registered handset:

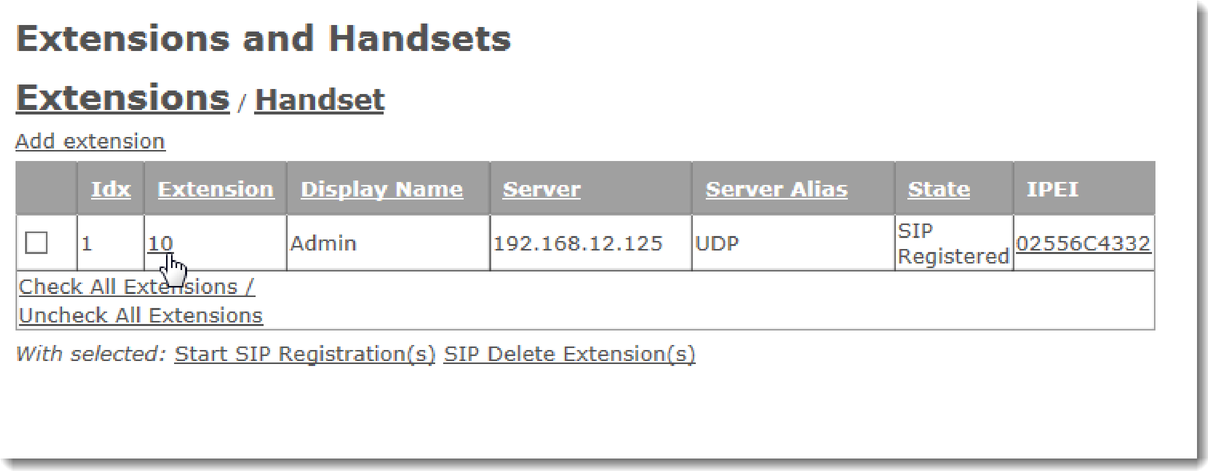
1. Log in to the administration interface of the Snom base station. See the official Snom wiki for more details.

2. Key in `admin` as the username and a password.

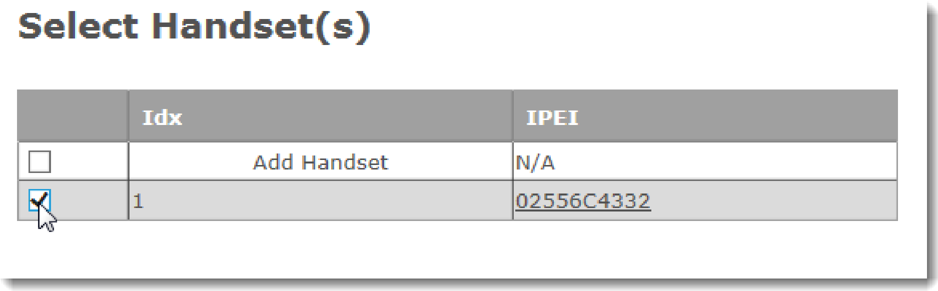
3. Click **OK**.

4. In the administration interface, go to **Extensions**.

5. In the **Extensions** table, click the number of the extension you want to assign.



6. In the Select **Handset(s)** table, select the handset you want to assign that extension to.



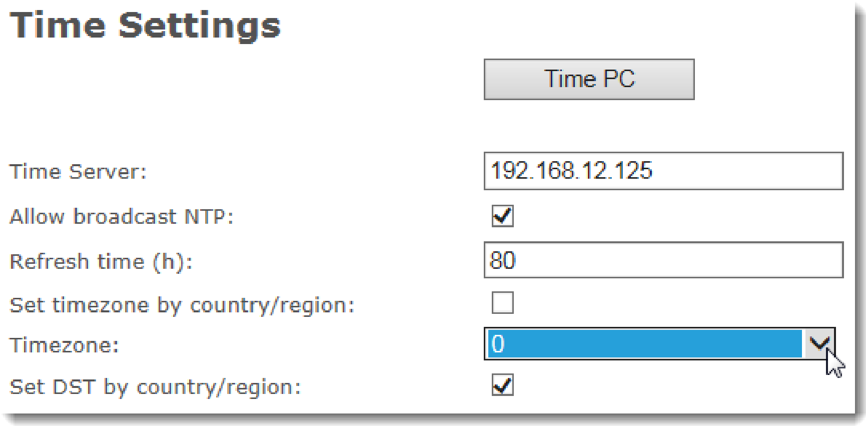
7. Click **Save**.

The display name and the number of the internal extension display on the main screen of the handset. Make a test call to verify the configuration of the handset.

Configuring time zones

To display your local time on handsets, you must configure the time zone in the administration interface of the base station:

1. Log in to the administration interface of the Snom base station.
2. Go to **Timezone**.
3. Disable the **Set timezone by country/region** option.
4. Set the **Timezone**.



5. Click **Save**.

Your handsets now automatically display the local time.

3.1.5 Configuring the Aastra 6755i IP Phone with Kerio Operator

This topic covers the basic configuration and usage of an Aastra 6755i IP phone with Kerio Operator.

> **NOTE**
> As of Kerio Operator version 2.3, this phone model can be automatically provisioned. Refer to the system requirements for a complete list of auto-provisioned phones.

Provisioning a new line (SIP account)

To configure a new line, log into the web administration of the phone by typing the phone's IP address into your browser. The default login for Aastra phones is `admin/22222`.

Navigate to **Advanced Settings > Line 1** and input the following values:

» Screen Name - The name that will appear on the phone's LCD.

» Phone Number - Your extension.

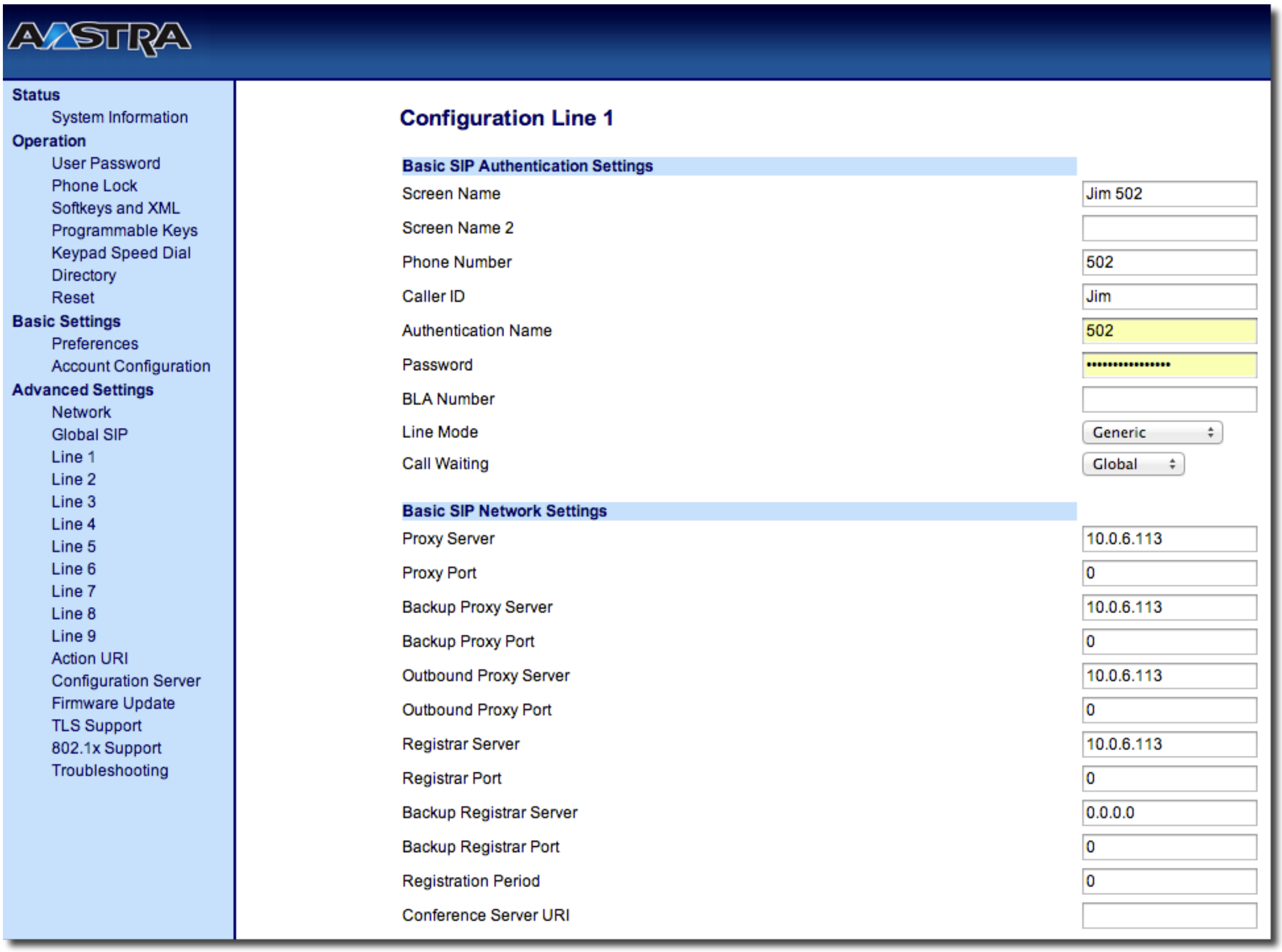
» Authentication Name - Your SIP login name (e.g. 230p1).

» Password - Your SIP password.

> **NOTE**
> Before configuring the line, make sure you've created an extension in Kerio Operator.

» Proxy Server - The IP Address or hostname of Kerio Operator.

» Registrar Server - The IP Address or hostname of Kerio Operator.

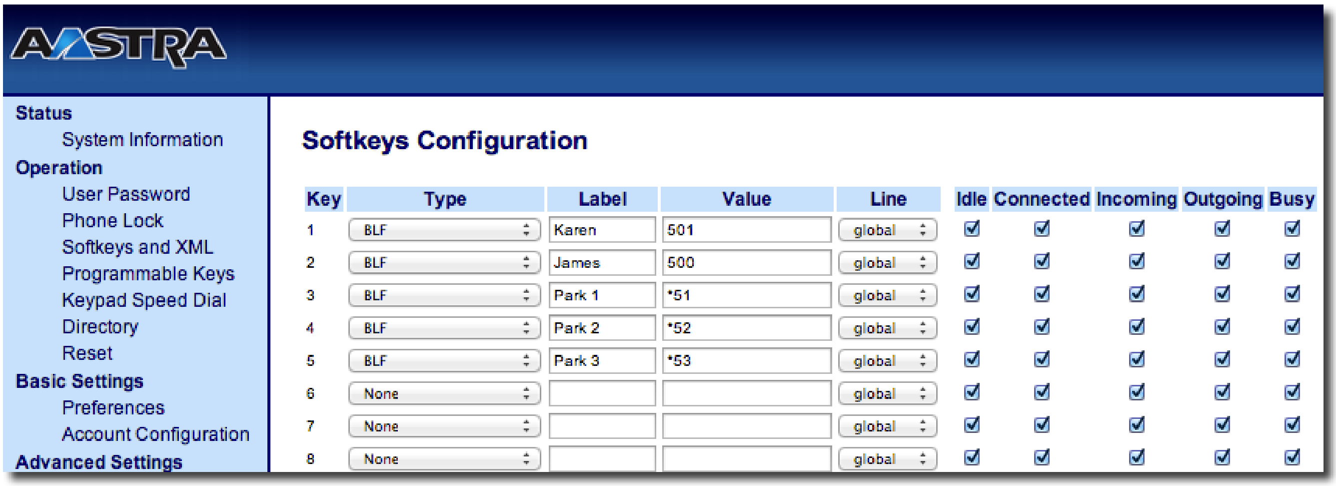


Configuring Busy Lamp Field (BLF)

Busy Lamp Field (BLF) allows the phone operator to view the status of other extensions, using an LED indicator. There is also an associated button next to the indicator, which acts as a speed dial to the extension being monitored.

The Aastra 6755i includes 12 programmable buttons / indicators that can be used for various functions, including BLF. The actions for these buttons are located under **Softkeys and XML**, and **Programmable Keys**. Both configuration dialogs offer the same functionality, however the softkeys label and accessibility can change based on the phone's status, while the labels of the programmable keys are static.

To set a button for BLF, select BLF from the pull down menu under the type column. The value will be the extension you want to monitor. In case of call parking, this will be any one of the assigned parking slots (e.g. *51).



Parking a call

Call parking is a feature that allows you to transfer a call to a temporary holding extension, which can then be joined from another phone. Before anyone can park a call, a range of parking slots must be defined in Kerio Operator. For configuration details refer to the KB topic Configuring and using call parking. To park an active call, press the Xfer (transfer) soft key. Then press the button which has been designated as a parking slot. Wait until you've been connected to the parking slot, then press again the Xfer button. The caller is then put into the parking slot, and will hear hold music. You can then hang up the phone.

> **NOTE**
> If nobody picks up the call within the defined timeout period (40 seconds by default), the caller will be bounced back to the person who parked the call.

Joining a parked call

If you have configured BLF with a parking slot, the LED will indicate that there is a parked call on that parking slot. To join the call, simply press the button and the phone will speed dial to the extension of that parking slot.

3.1.6 How to configure Busy Lamp Field (BLF) on Cisco SPA500S

Busy Lamp Field (BLF) feature allows users to monitor several other extensions. Sometimes, the term "Direct Station Selection" is used for the same functionality. State of the monitored extensions is usually indicated by a series of LED lights with buttons.

1. Idle state – LED is green

2. Error state – LED is orange

3. Ringing state – LED is red (blinking)

4. Busy or Connected state – LED is red

The "Call Pickup" function is usually configured together with BLF allowing, for example, a receptionist to pickup ringing calls with a press of a single button.

Kerio Operator configuration

Busy Lamp Field

There is no configuration needed to make BLF work.

Call Pickup

Enable the Call Pickup feature and configure an extension (e.g. "**"). When dialing "**10" while extension 10 is ringing, the call will be redirected to your phone.

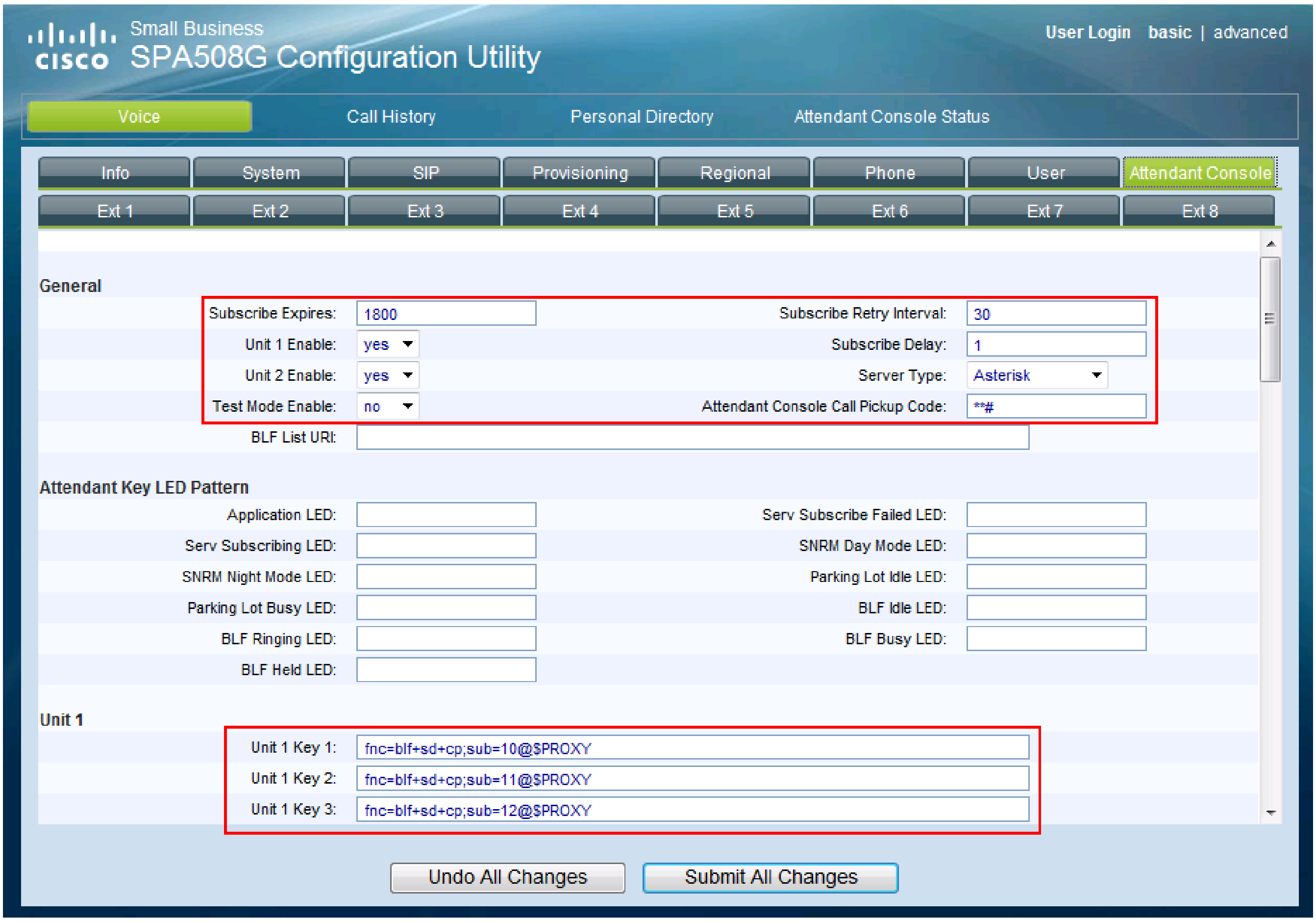
Phone configuration

This guide has been tested on a Cisco IP Phone SPA508G (firmware 7.4.8a) with an attendant console SPA500S (hw 1.0.6, sw 2.0.2).

1. Open the phone administration in your browser (eg. http://192.168.1.10).

2. Login as administrator and open advanced configuration.

3. Select the **Attendant Console** screen.



4. Set **Subscribe Expires** to `1800`.

5. Set **Unit 1 Enable** to `yes`.

6. Set **Unit 2 Enable** to `yes` if you have a second SPA500S unit.

7. Set **Server Type** to `Asterisk`.

8. Set **Attendant Console Call Pickup Code** to the extension of Directed Call Pickup followed by a # sign. When pressing a button, the # sing will be replaced by the extension number. Use "**#" for Directed Call Pickup at "**".

9. Configure the unit keys. Use `fnc=blf+sd+cp;sub=10@$PROXY` to monitor extension 10.

10. Don't forget to save the configuration.

Call Parking

To park a call, use the xfer (transfer) soft key. Then dial to a designated parking slot (e.g. *53). If you have configured BLF keys to monitor parking slots, you can simply press the key which is monitoring the parking slot (e.g. *53). After you hear the announcement you will then hear the hold music. You can then press the xferLx (blind transfer) soft key to join the caller into the parking slot. The caller is then parked, and you can hang up the call.

3.1.7 How to configure Busy Lamp Field (BLF) on snom phones

Busy Lamp field (BLF) feature allows users to monitor several other extensions. Sometimes the term *Direct Station Selection* is used for the same functionality. State of the monitored extensions is usually indicated by a series of LED lights with buttons.

- Idle state – LED is off.
- Ringing state – LED is blinking.
- Busy or Connected state – LED is on.

The **Call Pickup** function is usually configured together with BLF allowing, for example, a receptionist to pickup ringing calls with a press of a single button.

Operator configuration

Busy Lamp Field

There is no configuration needed to make BLF work.

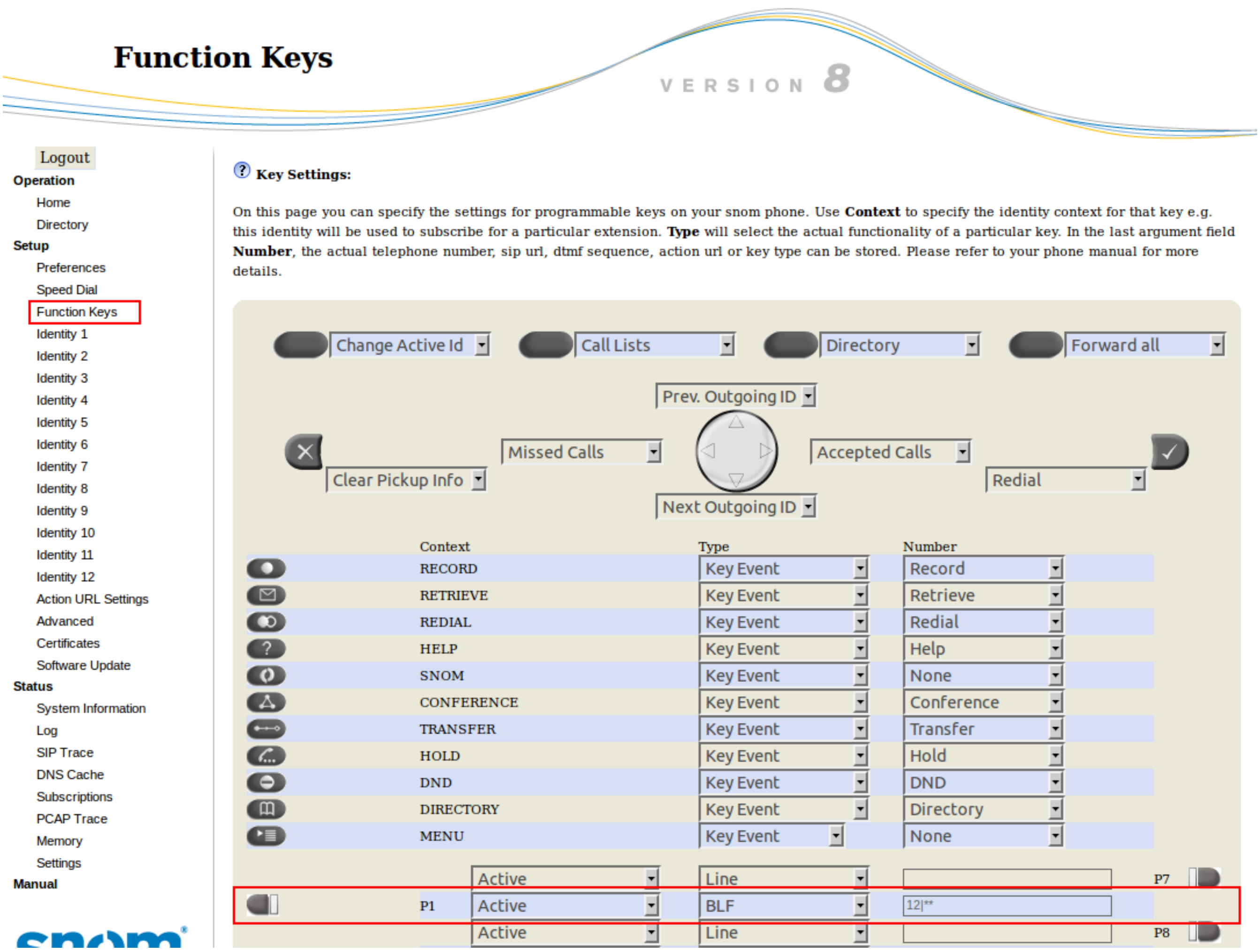
Call Pickup

Enable the Call Pickup feature and configure an extension (e.g. "**"). When dialing `**10` while extension 10 is ringing, the call will be redirected to your phone.

Phone configuration

Follow these steps on snom 360 (firmware 8.4.18) and snom 820 (firmware 8.4.32).

1. Open phone administration in your browser (eg. http://192.168.1.10).
2. Make sure that the phone is running firmware version 8.
3. View screen **Setup > Function Keys**.



4. Configure BLF function on the buttons P1...Px and set:

a. First field to `Active`

b. Second field to `BLF`

c. Third field to `<sip:extension@ipAddress;user=phone>|callPickup`. Here

- extension is the monitored extension (e.g. 10).
- ipAddress is the IP address of Operator (e.g. 192.168.1.1).
- callPickup is the Call Pickup extension (optional).

For example, `<sip:10@192.168.1.1;user=phone>|**`. In this example, `**` signifies that extension `10` doesn't use the Call Pickup feature.

5. Don't forget to save the configuration.

Call Parking

To park a call, first place the caller on hold. Then dial to a designated parking slot (e.g. *53). If you have configured BLF keys to monitor parking slots, you can simply press the key which is monitoring the parking slot (e.g. *53). After you hear the announcement you will then hear the hold music. You can then press the transfer button to join the caller into the parking slot. The caller is then parked, and you can hang up the call.

3.1.8 How to configure Busy Lamp Field (BLF) on Well phones

Busy Lamp field (BLF) feature allows users to monitor several other extensions. Sometimes the term "Direct Station Selection" is used for the same functionality. State of the monitored extensions is usually indicated by a series of LED lights with buttons.

1. Idle state – LED is off

2. Ringing state – LED is blinking

3. Busy or Connected state – LED is on

The **Call Pickup** function is usually configured together with BLF allowing, for example, a receptionist to pickup ringing calls with a press of a single button.

Operator configuration

Busy Lamp Field

There is no configuration needed to make BLF work.

Call Pickup

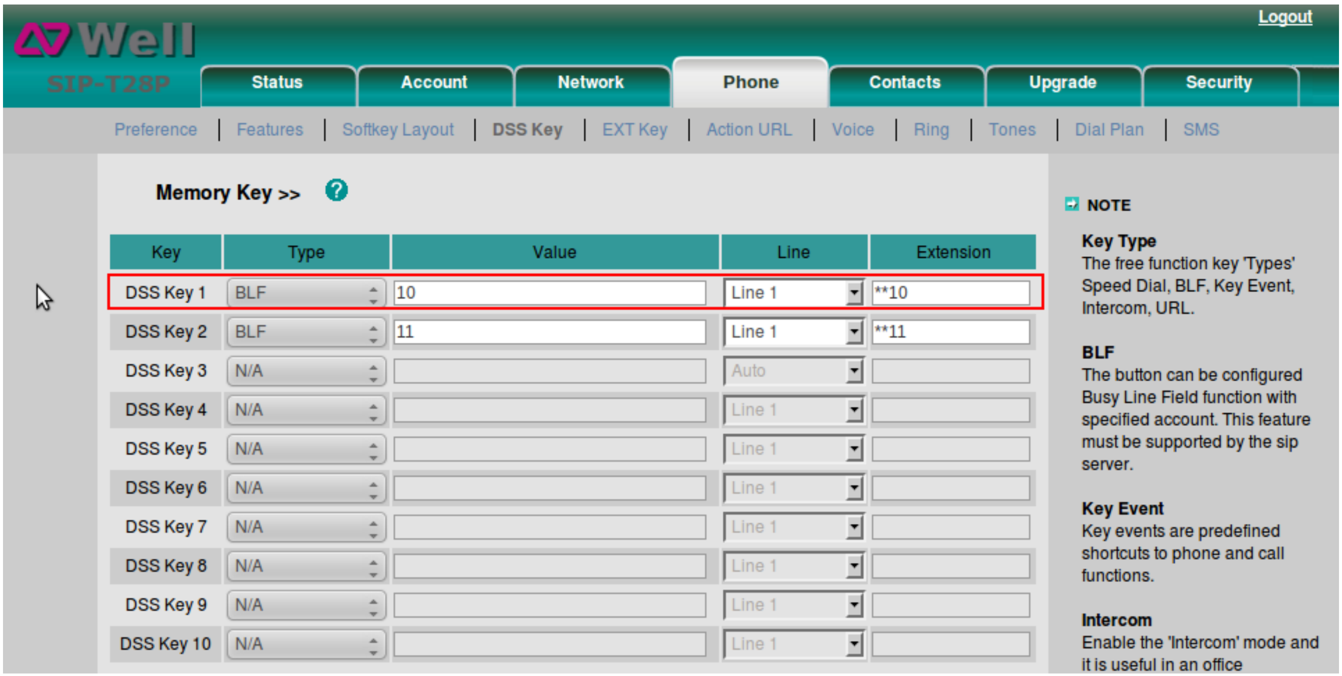
Enable the Call Pickup feature and configure an extension (e.g. "**"). When dialing "**10" while extension 10 is ringing, the call will be redirected to your phone.

Phone configuration

This guide has been tested on Well SIP-T28p (firmware 2.60.9.5).

1. Open phone administration in your browser (eg. Http://192.168.1.10).

2. View screen **Phone > DSS Key**.



3. Configure BLF function on the keys DSS Keys. Set:

a. Type to `BLF`.

b. Value the monitored extension (e.g. 10)

c. Extension to the Call Pickup extension + extension number. (e.g. **10). This field is optional.

4. Don't forget to save the configuration.

> **NOTE**
> It might take a while for phones to subscribe for notifications after Operator is rebooted. In this case either wait up to 10 minutes or reboot the phone.

3.1.9 Linksys/Cisco SPA: Setting the TFTP address without using the DHCP parameter 66

Use this topic, if:

» You want to use phone provisioning.

» Your DHCP server does not support parameter 66.

» You use the Linksys/Cisco SPA phone.

Setting the TFTP address without using the DHCP parameter 66 for Linksys SPA942

1. In a web browser, type the IP address assigned to the phone.

2. In the phone configuration, click **Admin Login > Advanced > Provisioning**.

3. Enter the TFTP address in **Profile Rule** for in the following format: `tftp://Kerio.Operator.IP.address/spa942$MA.cfg`

4. Click **Submit All Changes**.

5. After the phone restarts, restart the phone manually again. In the web browser, type the following address: `http://phone.IP.address/admin/reboot`

Screenshot 17: Configuring the Linksys/Cisco SPA phone

Profile rules for other Linksys/Cisco SPA telephones

Other Linksys/Cisco SPA telephones can have different administration interfaces, however, option **Profile Rule** is available in all telephones.

We prepared a list of profile rules for you:

Type	Profile Rule
Cisco 7940,7960	N/A
Cisco 7941,7961,7960G,7940G	N/A
Cisco SPA112	tftp://<ip>/_spa112<mac>.cfg
Cisco SPA122	tftp://<ip>/_spa122<mac>.cfg
Cisco SPA301	tftp://<ip>/Cisco/SPA301/<mac>.cfg
Cisco SPA303	tftp://<ip>/Cisco/SPA303/<mac>.cfg
Cisco SPA3102	tftp://<ip>/_spa3102<mac>.cfg
Cisco SPA501G	tftp://<ip>/Cisco/SPA501G/<mac>.cfg
Cisco SPA502G	tftp://<ip>/spa504G<mac>.cfg

Type	Profile Rule
Cisco SPA504G	`tftp://<ip>/Cisco/SPA504G/<mac>.cfg`
Cisco SPA508G	`tftp://<ip>/Cisco/SPA508G/<mac>.cfg`
Cisco SPA509G	`tftp://<ip>/Cisco/SPA509G/<mac>.cfg`
Cisco SPA525G	`tftp://<ip>/spa525G<mac>.cfg`
Cisco SPA525G2	`tftp://<ip>/Cisco/SPA525G2/<mac>.cfg`
Linksys PAP2T	`tftp://<ip>/_pap2t<mac>.cfg`
Linksys SPA1001	`tftp://<ip>/Linksys/SPA1001/<mac>.cfg`
Linksys SPA901	`tftp://<ip>/spa901<mac>.cfg`
Linksys SPA921	`tftp://<ip>/spa921<mac>.cfg`
Linksys SPA922	`tftp://<ip>/spa922<mac>.cfg`
Linksys SPA941	`tftp://<ip>/spa941<mac>.cfg`
Linksys SPA942	`tftp://<ip>/spa942<mac>.cfg`
Linksys SPA962	`tftp://<ip>/spa962<mac>.cfg`

3.2 Backups

This section provides information about server backup and data recovery.

3.2.1 Saving Kerio Operator configuration to MyKerio

> **NOTE**
> This information is designed for Kerio Operator 2.5 and newer.

Kerio Operator can automatically back up and upload the configuration files to MyKerio every day.

Each backup can include:

- Configuration files
- Local voicemail data
- System logs
- Call history log
- License
- Recorded calls
- Custom provisioning files /var/tftp

To configure backup to an FTP server instead, read the Saving configuration to FTP server topic.

Saving configuration to MyKerio

Before you start, connect your Kerio Operator to MyKerio. For details, read Adding Kerio Operator to MyKerio.

Once Kerio Operator is connected to MyKerio:

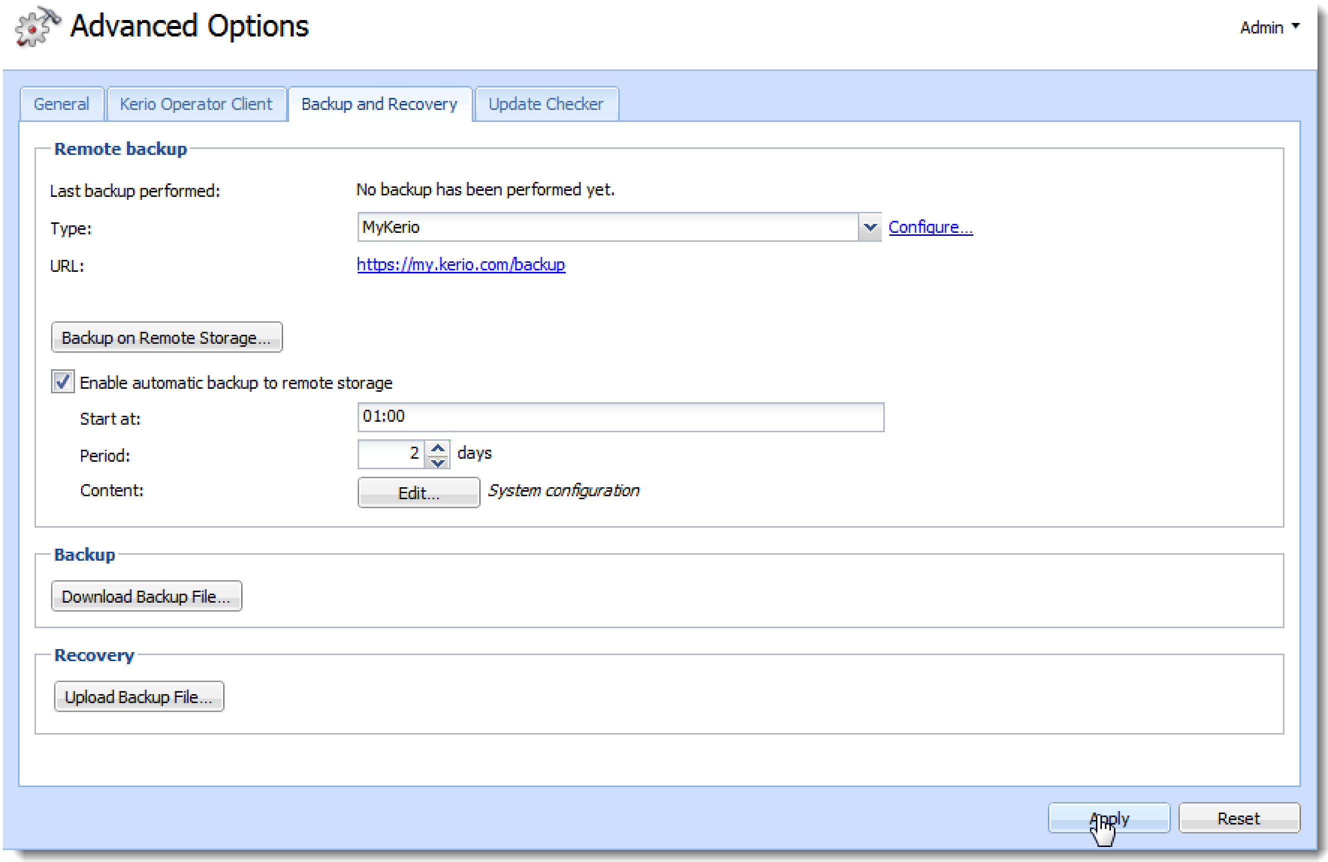
1. In the administration interface, go to **Advanced Options > Backup and Recovery > Remote Backup**.

2. In **Type**, select **MyKerio**.

3. Select **Enable automatic daily backup**.

4. Set the starting time and the period.

5. Click **Apply**.



Kerio Operator uploads configuration files once a day.

Restoring configuration from a backup

To learn how to restore your configuration from a backup, read the Backups in MyKerio topic.

3.2.2 Saving Kerio Operator configuration to FTP/SFTP or local storage

Kerio Operator can backup the following items:

- System configuration — system settings, IVR (auto attendant scripts), users, logos, firmwares etc.
- Local voicemail data — if you use integration with Kerio Connect, Kerio Operator sends voicemails via IMAP to Kerio Connect. These voicemails are not backed up.

- SSL certificate — only an active SSL certificate is backed up.
- System logs — all logs from the **Logs** section.
- Call history log — all logs from the **Status > Call History** section.
- License — a `.key` file with your licence.
- Recorded calls — locally saved recorded calls. You can also back up recorded calls to a FTP server.
- Custom provisioning files `/var/tftp`

Saving backups at a remote FTP/SFTP storage

1. Configure a FTP or SFTP server for storing the backup. For more information, refer to Configuring Remote FTP/SFTP Storage (page 304).

2. In the administration interface, go to **Advanced Options > Backup and Recovery**.

3. Under **Remote Backup,** select **Type** as either **FTP/SFTP**. In case of FTP/SFTP also set the **Number of files to keep**.

4. Verify the path to the remote storage type and test the settings using the **Backup on Remote Storage** button.

5. Check **Enable automatic backup to remote storage** to periodically perform automatic backups if required.

6. In the **Start at** field, specify the time of taking a backup.

7. In the **Period** field, specify how often backups should be performed.

8. Next to **Content**, click **Edit** and select content types for backup. By default, Kerio Operator backs up only a system configuration.

9. Save the settings.

Saving a backup locally

1. In the administration interface, go to **Advanced Options > Backup and Recovery**.

2. Under **Backup**, click **Download Backup File**.

3. Select a backup content. By default, Kerio Operator creates a full backup.

4. Click **Create Backup for Download**.

5. Click **Download** and save the file.

Recovering data from a backup

1. In the administration interface, go to **Advanced Options > Backup and Recovery**.

2. Click **Upload Backup File**.

3. Select the file and upload the backup to Kerio Operator.

4. When the **Recovery** dialog box appears, select the configuration and data for recovery.

5. Click **Recovery**.

6. Click **OK** to restart.

After the restart, the backup recovery is complete.

> **NOTE**
> After restoring from a backup, restart your browser in order to log back into the administration interface.

3.3 CRM integration and desktop dialers

This section helps you integrate with CRM systems and desktop dialers.

3.3.1 Salesforce integration with Kerio Operator

NOTE

Lightning Experience from Salesforce doesn't allow the settings described below. Switch to Salesforce Classic to complete the configuration.

Kerio Operator App for Salesforce is based on Call Center. The Call Center is an application embedded in Salesforce and integrates Salesforce with Kerio Operator. For more information about Call Centers, go to https://help.salesforce.com/.

Kerio Operator App for Salesforce enables:

- Click-to-dial.
- Displaying contacts, accounts and leads during the call.
- Logging calls into Salesforce.

To use Kerio Operator App for Salesforce, install the application. You can download it from Kerio Operator administration interface.

Kerio Operator supports:

- Salesforce Enterprise Edition
- Salesforce Performance Edition
- Salesforce Unlimited Edition

This topic helps you to install and configure Kerio Operator App for Salesforce. For more information, refer to Using Kerio Operator App for Salesforce (page 150).

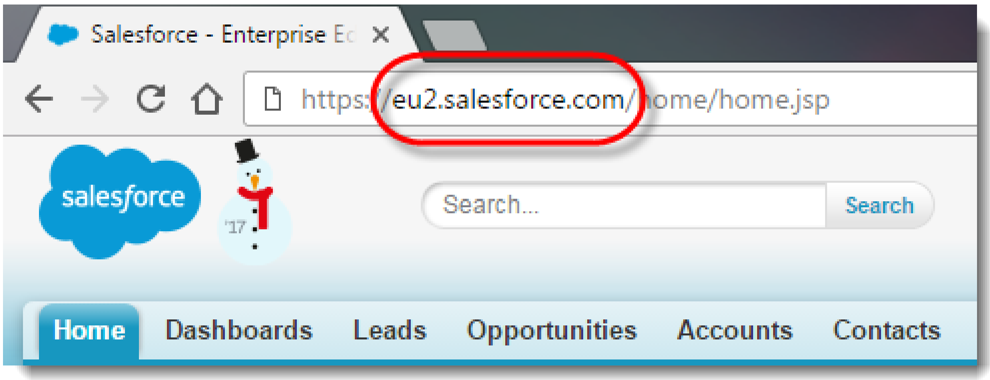
Configuring Kerio Operator

Add the Salesforce hostname to Kerio Operator and download **Call Center Definition** for Salesforce.

1. In the administration interface, go to **Integration**.

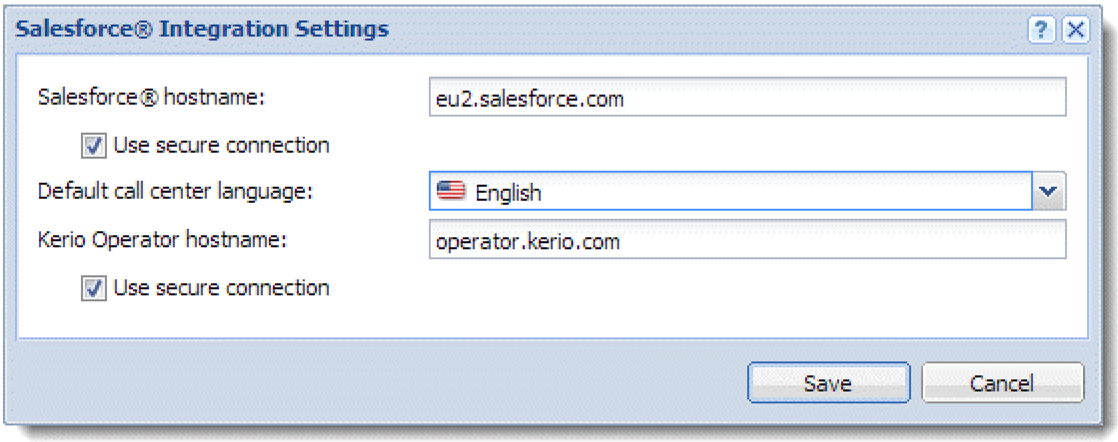
2. In the **Salesforce integration** section, click **Configure**.

3. Login to your Salesforce and copy the Salesforce hostname. Paste the hostname to **Salesforce hostname** in Kerio Operator.



4. Check if the Kerio Operator's hostname is complete. If the field is empty, type a correct Kerio Operator's hostname.

5. Save the settings.



6. Click **Download Call Center Definition**.

> **NOTE**
> The communication is based on HTTPS by default. Verify that port 443 is open in both directions and make sure that the hostname of the SSL certificate matches the Kerio Operator hostname. For more information, refer to Configuring SSL certificates (page 257).

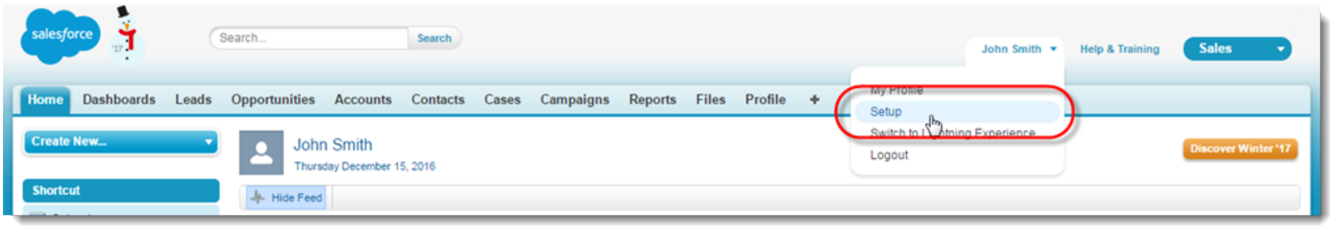
Configuring salesforce.com

Configuration is divided into three steps:

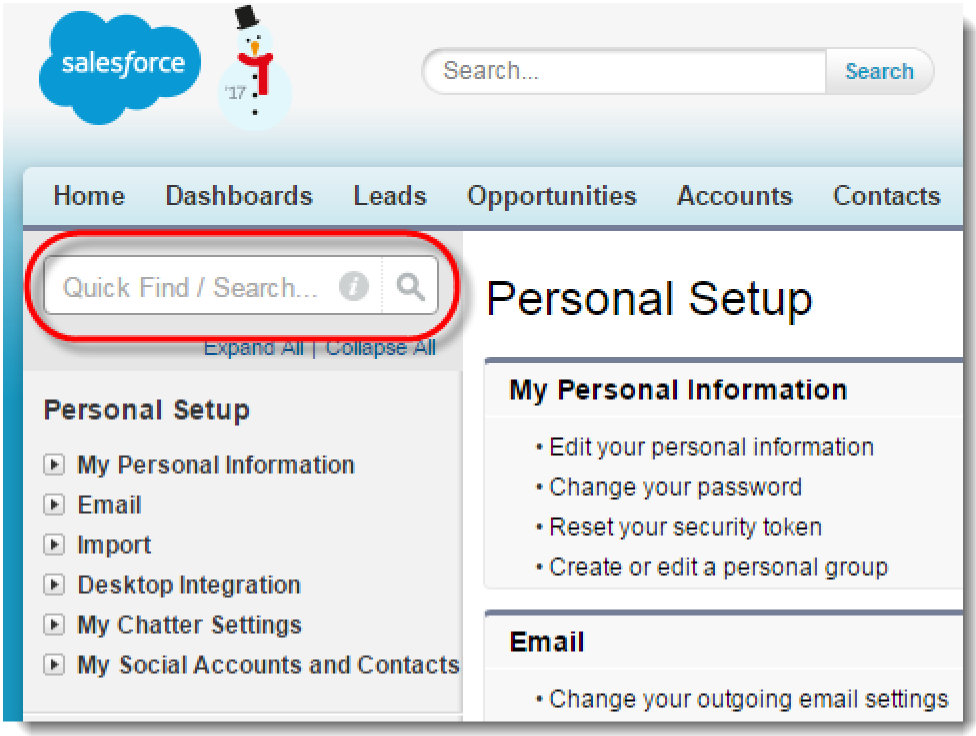
Step 1: Adding Kerio Operator Call Center

To add Kerio Operator Call Center to Salesforce, follow these steps:

1. In Salesforce, click your name and go to **Setup**.



2. In the **Quick Find**, type **Call Center** and click **Call Centers** in the results.



3. Skip the help page if it appears.

4. Click the **Import** button in the **All Call Centers** page.

5. Click the **Choose File** button and select the call center definition file you downloaded earlier.

6. Click **Import**.

Kerio Operator Call Center (Kerio Operator App for Salesforce) is installed in Salesforce. Now add users to the call center.

Step 2: Adding users to the call center

To add users (your colleagues) from Salesforce to Kerio Operator Call Center, follow these steps:

1. In **Kerio Operator Call Center**, click **Manage Call Center Users**.

2. Click **Add More Users**.

3. Leave the form as it is and click **Find**.

4. Select users and click **Add to Call Center**.

The users appear in the **Kerio Operator Call Center: Manage Users** table.

Go to **Home** in the main menu. You can see the Kerio Operator Call Center application if your user account is added in the Kerio Operator Call Center.

Step 3: Installing the Kerio Operator Open CTI Package

Kerio Operator Open CTI Package enables searching salesforce contacts, accounts and leads in the Kerio Operator Call Center application.

1. Go to Salesforce.

2. In the address bar of your browser, add this string after your Salesforce hostname (in our case it is `https://eu2.-salesforce.com/`): `packaging/installPackage.apexp?p0=04tb0000000QG2n` ... The final result is similar to:
`https://eu2.-salesforce.com/packaging/installPackage.apexp?p0=04tb0000000QG2n`

3. A **Package Upgrade Details** page is opened.

4. On page **Package Upgrade Details**, click **Continue**.

5. On page **KerioOperatorOpenCti**, click **Next**.

6. Select **Grant access to all users** and click **Next**.

7. Click **Install**.

8. If you are successful, the application answers that the installation is complete.

You can test all features of Kerio Operator App for Salesforce. For more information, refer to Using Kerio Operator App for Salesforce (page 150).

Configuring number transformation for calls from Salesforce

To make calling via Kerio Operator App for Salesforce easy, add number transformations which ensure that numbers are dialed correctly from Salesforce.

For more information, refer to Using number transformation (page 205).

Configuring outgoing prefixes

You can also configure prefixes in Kerio Operator Call Center. However, number transformation is recommended.

1. Go to Kerio Operator Call Center.

2. Click **Edit**.

3. Change prefixes in the **Dialing Options** section.

4. Click **Save**.

Prefixes are the same for Kerio Operator and Salesforce now.

3.3.2 Using Kerio Operator App for Salesforce

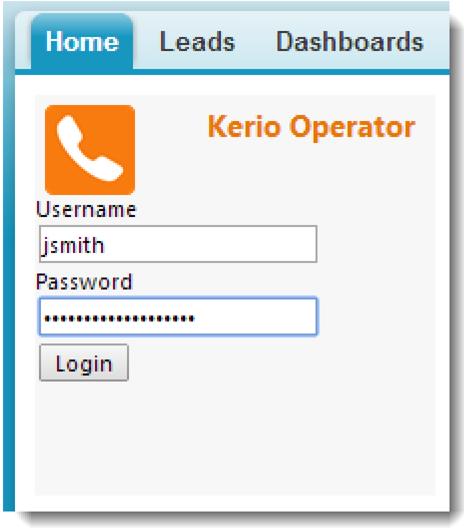
Learn how to use Kerio Operator App for Salesforce. If you need to install and configure Kerio Operator App for Salesforce, go to Salesforce integration with Kerio Operator .

Log into Kerio Operator App for Salesforce

Dialing of numbers is available to all users who use a software or hardware phone which has an extension of the Kerio Operator PBX configured. If the telephone is connected, it is possible to dial the called number using Kerio Operator App for Salesforce.

Before using the Kerio Operator App for Salesforce, you will be prompted to login:

1. Login to Salesforce, go to **Home**. You can see Kerio Operator App for Salesforce.

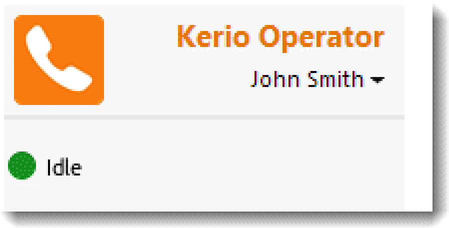


Screenshot 18: Kerio Operator App for Salesforce on Homepage

2. To login to Kerio Operator App for Salesforce, type Kerio Operator credentials (the same credentials as for Kerio Phone).

3. Click **Login**.

If you succeed, the application is open and the extension is idle.



Screenshot 19: A successfull login with a registered extension

If your extension is offline, you have not registered phone extension. You should do the following:

» If you have more extensions, change the extension (click your name and select the extension).

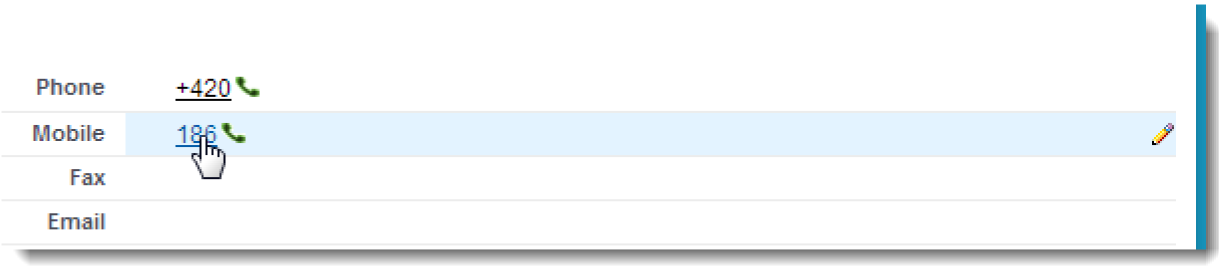
» Check that your phone is working.

» Contact your system administrator.

Dialing calls (click-to-dial) from Salesforce

To dial the number, click any number marked as a phone number.

Dialing in Salesforce works on a callback basis. This means that Kerio Operator App for Salesforce connects directly with the PBX and the PBX contacts back your phone. Therefore, the side effect of this operation is that upon clicking on **Dial**, your phone starts to ring as well as the called person's one. Pick it up and wait for the called person to answer.

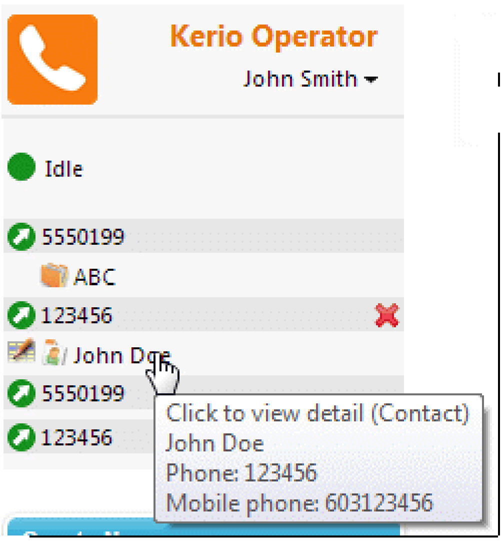


Screenshot 20: The phone number

> **NOTE**
> If the phone icon is gray, used extension is offline. For more information, refer to Log into Kerio Operator App for Salesforce (page 150).

Displaying contact, account or lead during the call

Kerio Operator App for Salesforce can log all incoming and outgoing external calls. The call history (last five calls by default) appears directly in Kerio Operator App for Salesforce. When you click on an item connected with the phone number (it can be an account, contact or lead), the item appears and you can see the details of the caller.

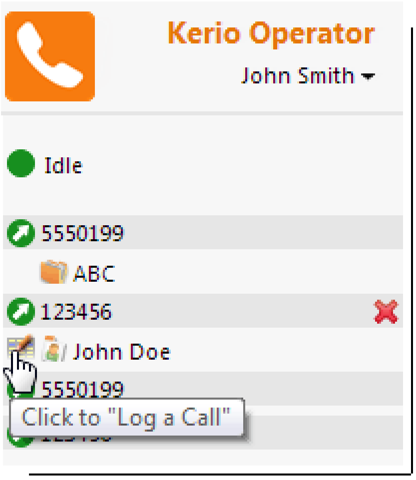


Screenshot 21: Click to view detail

Call logging in Salesforce

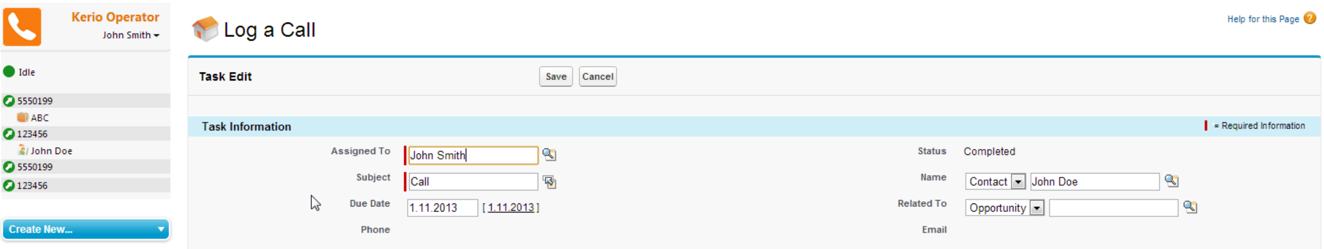
You can log incoming and outgoing calls to the **Activity History** in Salesforce.

1. In the Kerio Operator App for Salesforce, move your cursor onto contact, account or lead.



Screenshot 22: Click to Log a call

2. When a text **Click to "Log a Call"** appears, left-click. The **Log a Call** page appears.



Screenshot 23: Log a Call dialog

You can edit a task immediately.

Advanced settings

Changing prefixes

If you need to change a prefix, click your name and select **Dial out prefix**.

If you are required to dial a prefix for outgoing calls, you can configure the Kerio Operator App for Salesforce to automatically prepend dialed numbers with a prefix. If you need to assign a prefix, click your name and select dial out prefix.

Changing a language

If you need to change a language of Kerio Operator App for Salesforce, click your name and select **Change language**.

Opening Kerio Phone from Kerio Operator App

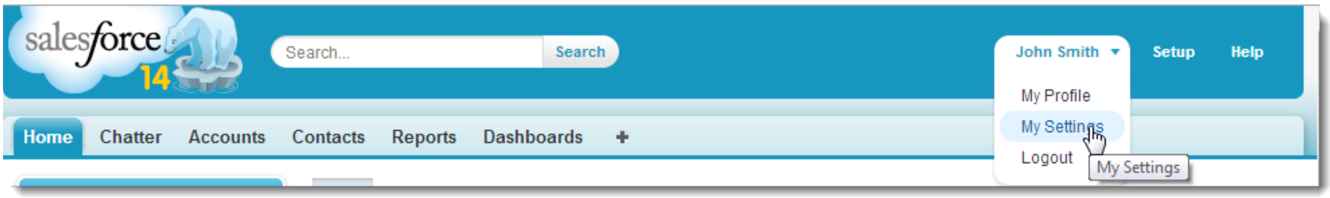
If you want to open Kerio Phone directly from Kerio Operator App, click your name and select **Open Kerio Operator Client**.

The advantage is that you do not have to fill the credentials twice.

Configuring a time zone

Salesforce time zone should be consistent with your computer:

1. In Salesforce, go to **My Settings**.



Screenshot 24: My Settings

2. Click **Personal**.

3. Click **Language & Time Zone**.

4. Change the time zone and save the settings.

3.3.3 Configuring OutCALL for dialing from the Microsoft Outlook contacts

OutCALL allows you to dial calls directly from Microsoft Outlook 2000 and newer. It uses the AMI interface (Asterisk Manager Interface) which Kerio Operator supports.

Download OutCALL at http://outcall.sourceforge.net/

What you need

- Microsoft Outlook 2000 or newer.

> **WARNING**
> OutCALL supports only the 32-bit version of Microsoft Outlook.

- Password for dialer (AMI) generated in Kerio Operator.
- Install and configure OutCALL on the user's computer.

Settings in Kerio Operator

Read topic CRM integration using AMI for information on Kerio Connect settings. The standard settings are as follows:

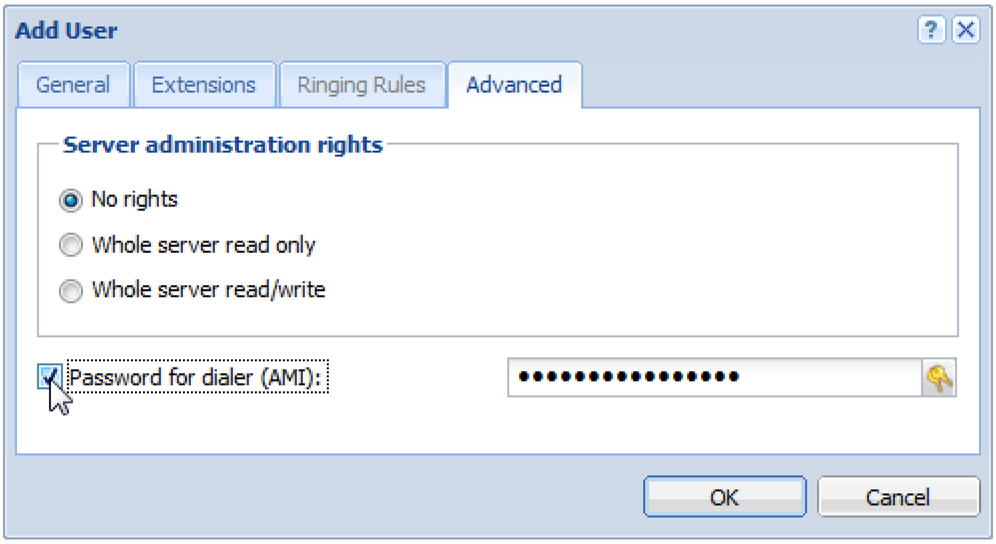
1. Login to Kerio Operator as an administrator.

2. Open the **Configuration > Users** section.

3. Double-click the user whom you wish to enable the OutCALL communication.

4. This opens the **Edit user** dialog. In the event, go to the **Advanced** tab.

5. Check **Password for dialer (AMI)** and copy the password (displayed upon clicking on the icon with keys).



Screenshot 25: Dialer Password

Configuring OutCALL

1. On user's computer, install Microsoft Outlook and create a mail account (unless it has been created before).

2. Close Microsoft Outlook before installing OutCALL.

3. Download OutCALL at http://outcall.sourceforge.net/.

4. Install it on user's computer.

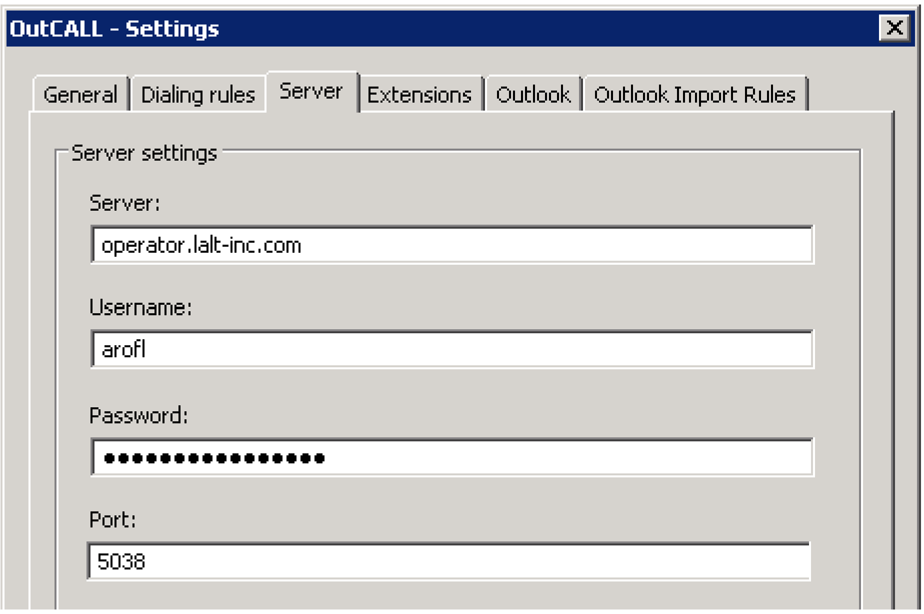
5. If the installation was successful, run OutCALL. OutCALL will run as a service with an icon displayed in the notification area (System Tray).

> **WARNING**
> For more information, refer to Troubleshooting (page 157).

6. Right-click the icon. Context menu is displayed.

7. Select **Settings**.

8. This opens the **OutCALL — Settings** dialog. Go to the **Server** tab.



Screenshot 26: OutCALL — configuring connection to Kerio Operator

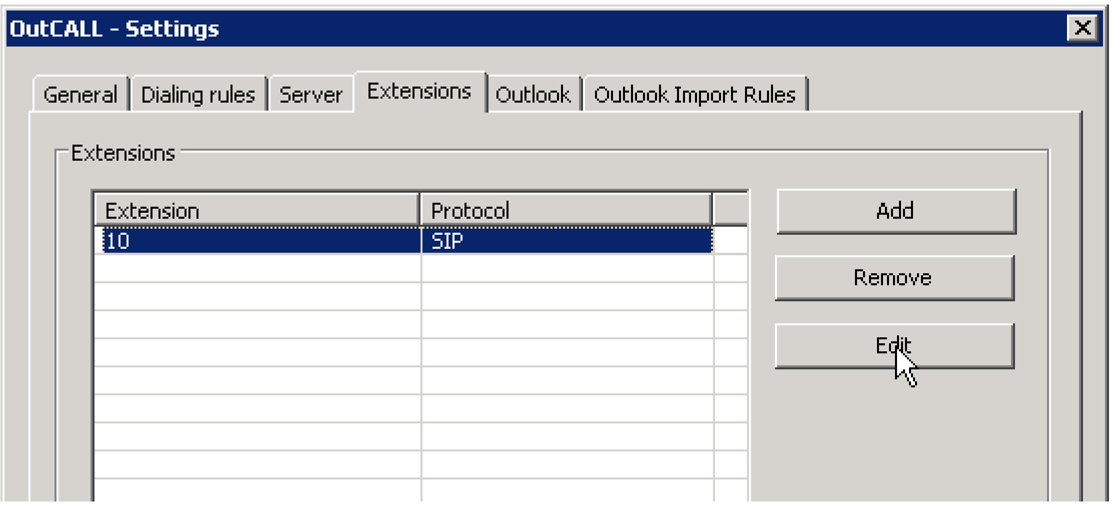
9. In the **Server** field, enter the DNS name or IP address of Kerio Operator.

10. In the **Username** field, enter the username in the same format as it is used in Kerio Operator.

11. In the **Password** field, enter password for dialer (AMI).

> **WARNING**
> Password for dialer does not equal the username used to login to Kerio Phone.

12. Switch to the **Extensions** tab. The default extension is set to 10. Click on **Edit** and enter the SIP username (the SIP username format can be found in the **Kerio Operator** administration in section **Extensions**).



Screenshot 27: OutCALL — setting user extensions

13. Save the settings. Once the application connects to Kerio Operator, a pop up window is displayed informing about the successful connection.

Troubleshooting

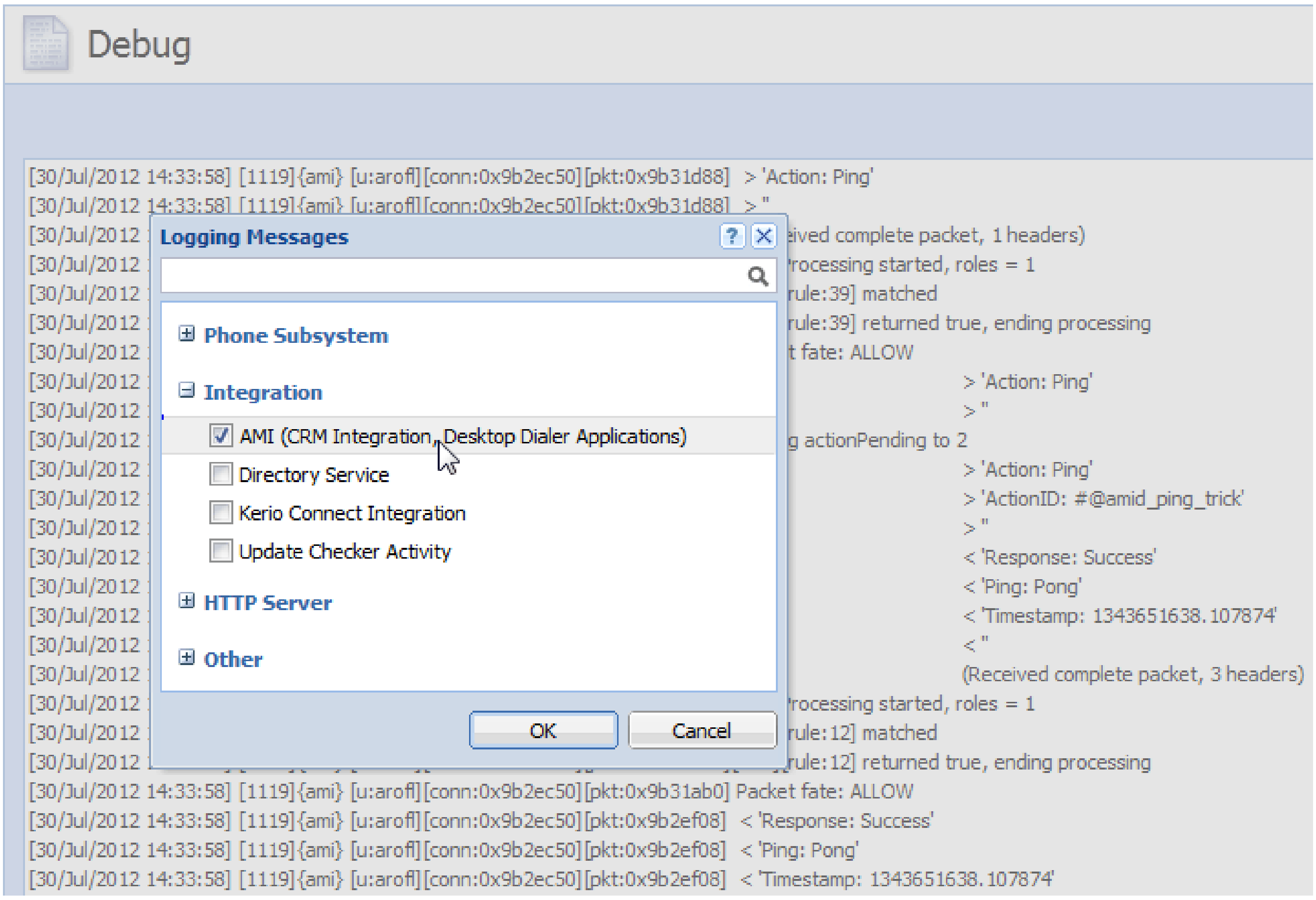
OutCALL stops working after the start

» OutCALL stops working after the start if the path to the folder contains special characters. OutCALL does not work on localized versions of Microsoft Windows XP where the Application Data folder is localized (e.g. in Czech, `"C:\Documents and Settings\user\Data aplikací"`). The problem is solved on Microsoft Vista and newer because the folder is not localized.

» OutCALL stops working after the start if the username used for login to the computer contains special characters. The solution is easy: change the username so that it does not contain special characters.

Logs

If you wish to verify the communication between OutCALL and Kerio Operator, look at the logs:



Screenshot 28: The Debug log — AMI

1. Login to Kerio Operator administration.

2. Go to section **Logs > Debug**

3. Right-click the log area and select **Messages**.

4. This opens the **Logging messages** dialog window. Check the **AMI (CRM Integration, Desktop Dialer Applications)** option.

3.3.4 CRM integration using the AMI

Asterisk Manager Interface (AMI) is an interface which enables other applications to connect to Kerio Operator (which includes Asterisk) and to communicate via the AMI commands. You can use it to make phone calls. It enables you to:

- dial calls from your CRM system,
- monitor call statuses in your CRM system (e.g., create logs),
- direct calls to another extension or terminate calls in your CRM system.

Connecting Kerio Operator with other applications

You can connect an application with Kerio Operator very easily. The settings are different for connections with a client (the server-to-client connection) and with a server (the server-to-server connection).

How to connect a client application (desktop application for dialing numbers) with Kerio Operator

To connect the applications, you need the username and password of the client application user:

1. In the administration interface, go to **Configuration > Users**.
2. Select a user and open the **Edit User** dialog.
3. Go to tab **Advanced** and check option **Password for dialer (AMI)**.
4. Click on the icon and note down the displayed password.
5. Enter the username and password in the client application to authenticate.

How to connect a server (CRM system) with Kerio Operator

You need the authentication data which you enter to your CRM system:

1. In the administration interface, go to **Configuration > Integration > General**.
2. Click **Configure** at **Third party CTI integration (AMI)**.
3. Check **Third party CTI integration is enabled**.
4. Click **Add**.
5. Enter **Account name** (usually the name of the CRM system).
6. The password is generated automatically. Click on the icon and note down the password.
7. To test the communication, set the permissions to full control. If the communication is successful, you may limit the permissions.

> **NOTE**
> Some applications allows you only to originate calls but they use asterisk commands which require a higher level of permission (usually full control).

8. Login to your CRM system and enter the password for the AMI integration.
9. Test the communication by dialing an extension.

Application we have tried and prepared a configuration guide

For more information, refer to Configuring OutCALL for dialing from the Microsoft Outlook contacts (page 154).

What to do when communication fails

Consult the logs in Kerio Operator:

1. In the administration interface, go to section **Logs** > **Debug**.

2. Right-click on the log screen and select option **Messages** in the context menu.

3. This opens the **Logging Messages** dialog box. Check the **AMI (CRM Integration, Desktop Dialer Applications)**.

Configure the internal firewall of Kerio Operator

1. In the administration interface, go to section **Configuration** > **Network** > **Firewall** and check the settings.

2. If your CRM system is located outside your local network, add its IP address in section **Configuration** > **Definitions** > **IP Address Groups**,

3. Go back to section **Configuration** > **Network** > **Firewall** and select a new IP address group for the integration with the CRM system.

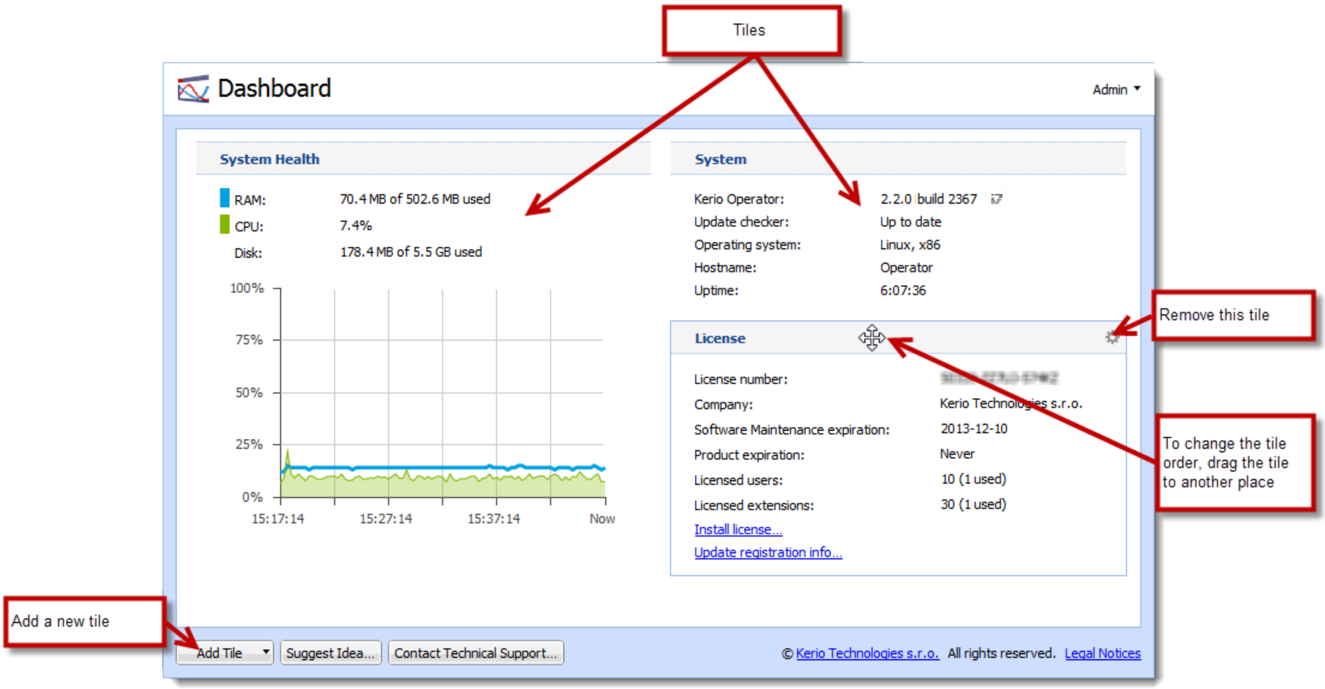
3.4 Monitoring

This section contains information about:

3.4.1 Using Dashboard in Kerio Operator

Kerio Operator includes a customizable Dashboard. A dashboard consists of tiles and each tile displays a different type of information (graphs, statistics, etc.). It gets displayed as you log into Kerio Operator.

It is also accessible from **Configuration** > **Dashboard**.



3.4.2 Monitoring Kerio Operator

When you are experiencing problems with your connection, we recommend you to monitor the status of your PBX.

Monitoring can be done using the **Status** section:

Monitoring active calls

All current calls can be viewed under **Status > Calls**.

You can see a table where each call occupies one line and a graph displays a number of calls in time in the **Calls** section.

Go to the **Calls** section, especially in case that you plan to restart the PBX which may result in an undesired termination of a call in progress.

Call History

The Call History section keeps a list of all internal and outbound calls of the PBX.

Call History can be viewed under **Status > Call History**.

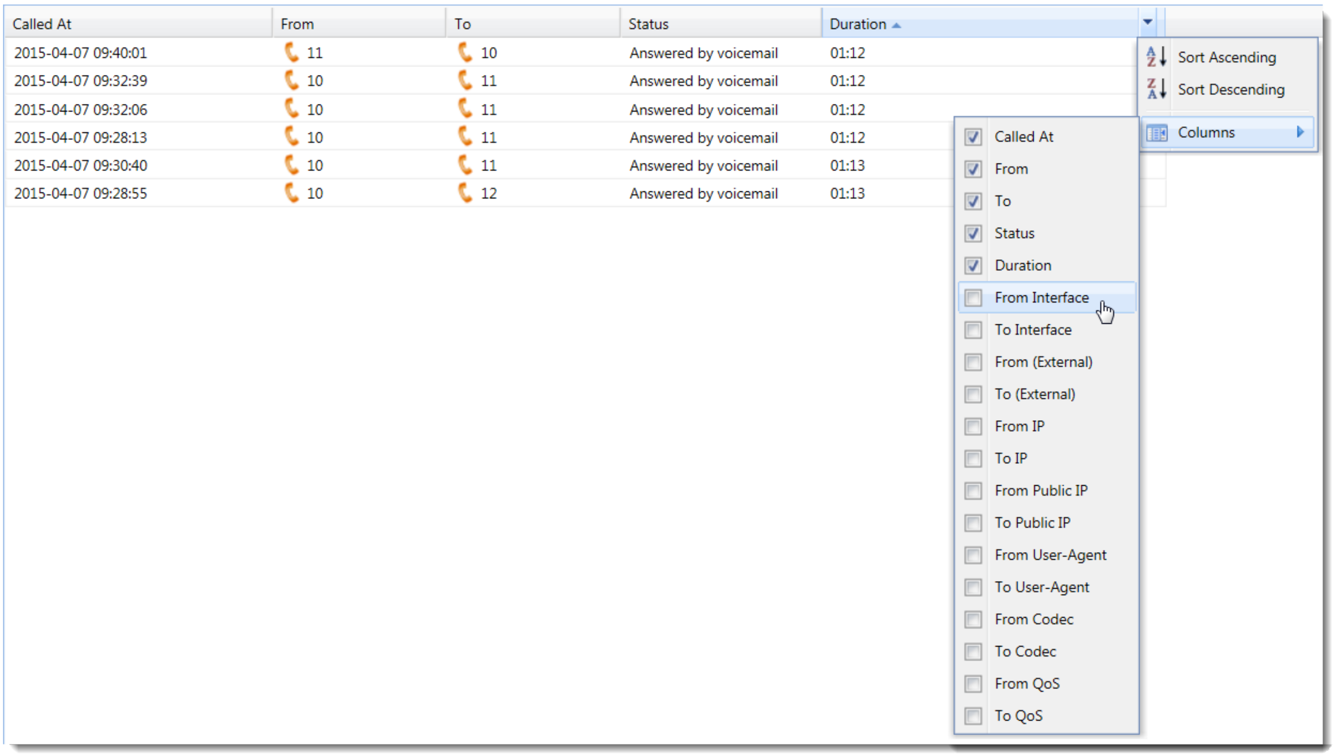
To add or remove columns in the call history:

1. In the administration interface, go to **Status > Call History**.

2. Mouse-over a name of a column and click the arrow on the right side.

3. In **Columns**, you can:

- select new columns to add them to the **Call History**,
- deselect columns to remove them.



Each line contains information about one call. The following actions can be applied to the call history:

Action	Description
Export to a CSV file	You can click on **Advanced >Export to a CSV file** to save the file on your local drive.
Clear	Click on **Advanced > Clear** and confirm your decision in the corresponding dialog.

NOTE
Individual users can delete their history in the **Kerio Operator Softphone**. However, this operation only hides the data. They are not removed from the PBX and logs.

Monitoring Recorded Calls

Section **Status > Recorded Calls** displays all calls recorded from call queues. This section displays a table where each recorded call occupies one row. Select a call to listen to it, download it to your computer or remove it.

Click **Settings** to record calls locally or to a remote storage. For more information, refer to Saving recorded calls (page 298).

Monitoring a Kerio Operator dial plan

A dial plan contains a list of all the used extensions and their users. You can export this list to a CSV file or print it.

Go to section **Status > Dial Plan** to see the list:

Export to CSV — the button exports the data in the format described in table.

Extension Number	Type ID	Description
111	1	Winston Smith

Extension Number	Type ID	Description
112	1	Ada Monroe
50	7	Voicemail

Changing the Dial Plan

If you use automatic phone provisioning and the change in your dial plan may affect automatically provisioned phones, update of the phones configuration is needed. Kerio Operator detects such changes automatically and displays a warning. If you confirm this warning, phones will be restarted at the time you selected in the dialog. You can restart the phones later manually in section **Provisioned Phones**. To restart the phones, click on the **Advanced > Restart all phones** button.

Monitoring active conferences

All current conferences can be viewed under **Status > Conferences**. The window displays two tables. Each line in the first table displays one conference. The second table displays information about individual conferences. Just select a conference and the details in the bottom table are updated.

Monitoring call queues

All active call queues and their parameters can be observed in section **Status > Call Queues**. The window displays three tables. Each line in the first table displays one call queue.

The other tables display agents and callers in a queue. Just select a queue and the details in table **Agents** and **Callers** are updated.

You can also reset the call queue statistics to start from zero. Use the **Reset** button.

System Health

The administration interface allows you to view the status of CPU, memory and disk space of your computer with Kerio Operator.

System status can be viewed under **Status > System Health**.

In this section, click **Tasks** to:

- restart telephony subsystem
- reboot Kerio Operator
- power off Kerio Operator
- do factory reset of Kerio Operator

The **Support information** link generates an asterisk configuration file and last 100 lines of all logs. This information may be helpful especially when solving issues in cooperation with the Kerio Technologies technical support.

See detailed information about disk space usage by clicking on **Details**. This opens a dialog with information about disk usage of audio files, voicemail and configuration file of Kerio Operator.

3.4.3 Managing logs in Kerio Operator

Logs are files where information about certain events (e.g. error and warning reports, debugging information) is recorded. Each item is represented by one row starting with a timestamp (date and time of the event). Messages in logs are displayed in English for every language version of Kerio Operator.

Configuring logs

Logs are available in the Kerio Operator administration interface in section **Logs**.

When you right-click in a log, you can configure the following settings (available in all logs):

Option	Description
Save log	You can save whole logs or a selected part in a `txt` or `HTML` format. See also **Log Settings** option.
Highlighting	You can save any part of text in logs for better reference. Specify a substring or regular expression and all rows containing such text will be highlighted.
Log Settings	Apart from immediate savings, you can configure regular saves of individual logs, specifying the size and number of saved files. You can also enable external logging to a Syslog server.
Clear Log	Use this option for deleting a log.

Types of logs

Type	Description
Auth	The **Auth** log includes information about all successful attempts to login to Kerio Operator (to the administration or client interfaces). Failed login attempts are logged into the **Security** log.
Config	The **Config** log stores the complete history of communication between Kerio Operator Administration and the server. It is possible to determine what administration tasks were performed by a specific user.
Debug	**Debug** log is a special log which can be used to monitor specific information. This is especially useful for problem-solving. To enable the **Debug** log, right-click in the log window and select the **Messages** option in the context menu. In the opened dialog window, select specific information you wish to monitor. **WARNING** In addition, displaying too much information slows Kerio Operator's performance. We recommend that you only display information that you are interested in and only when necessary.
Error	The **Error** log displays serious errors that affect the functionality of the entire PBX. The Kerio Operator administrator should check this log regularly and try to eliminate problems found here. Otherwise, users might have problems with some services or/and serious security problems might arise.
Event	The **Event** log gives information about phone and interface registrations, phone provisioning, new versions of Kerio Operator, etc.
Kernel	The **Kernel** log contains records generated by the operating system. It includes information about starting and stopping of the server, logs generated by individual processes, etc.
Security	The **Security** log contains the failed login attempts to Kerio Operator.
Warning	The **Warning** log shows error warnings which are not severe. Typical examples of such warnings are messages stating that a user with administrator rights has a blank password or that a user account of a given name does not exist. Events recalling warning messages in this log do not seriously affect the PBX functionality. However, they can point at current or possible problems. The **Warning** log can help if for example a user is complaining that services are not working.

3.4.4 SNMP monitoring

Enabling SNMP monitoring in Kerio Operator

SNMP is a protocol which allows you to monitor Kerio Operator status.

1. In the administration interface, go to **Network > General**.

2. Click **Configure**.

3. Check **SNMP monitoring is enabled**.

4. In the **Location** field, type any text which will help you recognize the server and its location.

5. In the **Contact** field, type address which will help you recognize the server and its location.

6. Select which version to use — **2c** (default password contains string `public`) or **3**.

> **NOTE**
> Version 2c supports passwords as plain text only (community string), version 3 supports encryption. Some monitoring tools, however, do not support version 3.

Using Cacti for SNMP monitoring

Cacti is a standard monitoring tool which can handle the SNMP protocol.

If you use Cacti to monitor your servers, go to cacti.net to acquire a template for Kerio Operator. It was created for Cacti 8.8 and newer and contains graphs similar to screen-shot below.

Graph Template

```
[success] ucd/net - CPU Usage [update]
[success] Unix - Ping Latency [update]
[success] ucd/net - Load Average [update]
[success] ucd/net - Memory Usage [update]
[success] Kerio Operator - Active Calls [update]
[success] Kerio Operator - Active Channels [update]
[success] Kerio Operator - Webserver Processes [update]
[success] Kerio Operator - Processed Calls [update]
[success] Kerio Operator - Reconfiguration [update]
[success] Kerio Operator - Asterisk Modules Count [update]
[success] Kerio Operator - Uptime [update]
[success] Kerio Operator - Processes [update]
[success] Interface - Traffic (bits/sec) [update]
[success] Interface - Errors/Discards [update]
[success] Interface - Unicast Packets [update]
[success] Interface - Non-Unicast Packets [update]
[success] Interface - Traffic (bytes/sec) [update]
[success] Interface - Traffic (bits/sec, 95th Percentile) [update]
[success] Interface - Traffic (bits/sec, Total Bandwidth) [update]
[success] Interface - Traffic (bytes/sec, Total Bandwidth) [update]
[success] ucd/net - Available Disk Space [update]
```

To import the template, follow these steps:

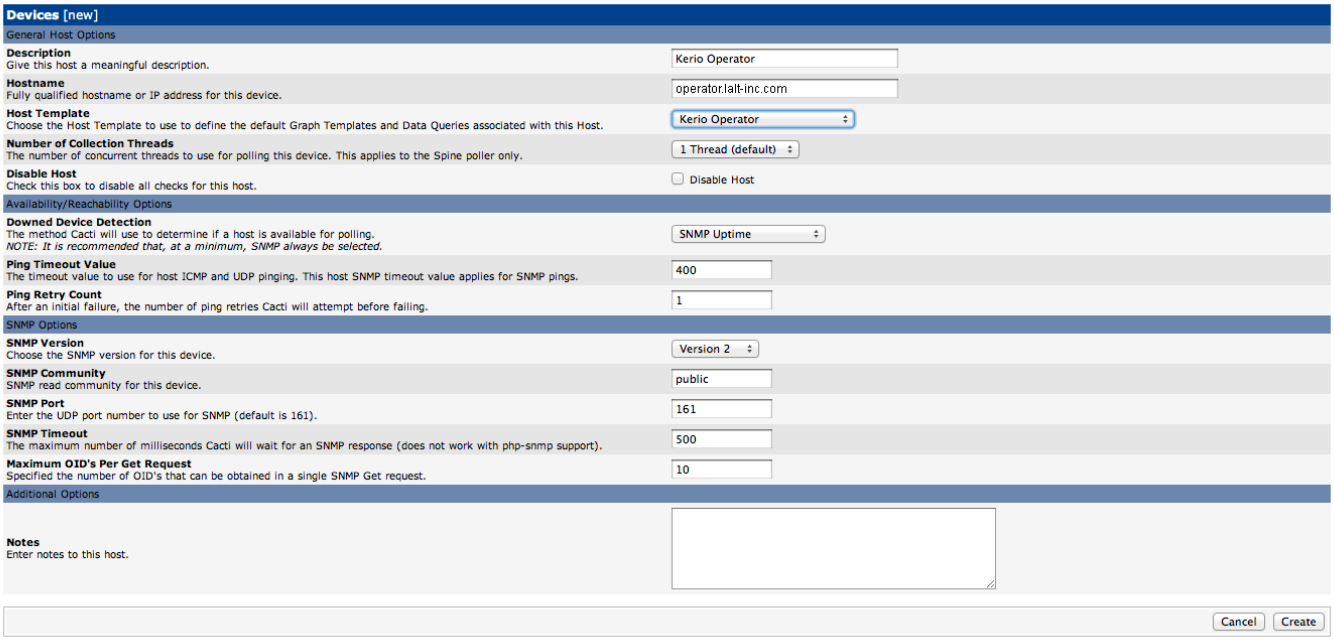
1. Connect to your Cacti with your browser.

2. Download the template archive and extract it.

3. Select **Import Templates**.

4. Click **Choose File** and select the XML template file.

Once the import is finished, go to the **Devices** section, add a new device, enter the DNS name of Kerio Operator and in the **Host Template** menu select Kerio Operator.



3.4.5 Monitoring active calls

NOTE

This information is designed for Kerio Operator 2.3 and newer.

Call monitoring allows you to participate in any active call by dialing a special prefix, followed by an extension.

You can use call monitoring in call centers where supervisors need to monitor trainees during conversations with customers.

WARNING

When you join an active call, the active callers have no indication that you have joined the call.

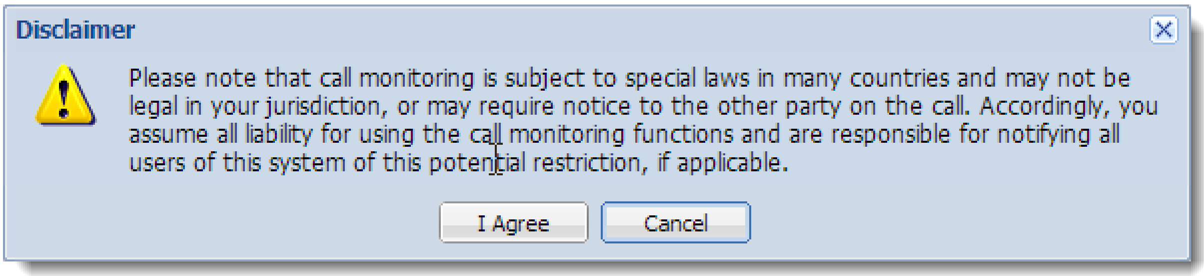
Call monitoring is protected by a PIN number. Whoever knows the PIN can listen to any extension in your telephony subsystem. Therefore, we recommend to set special call permissions for people who can use the call monitoring prefix.

The default prefix for call monitoring is * 6, and it is configured in the PBX services. The prefix is disabled by default and you have to enable it manually.

Configuring call monitoring

To configure the call monitoring service, follow these steps:

1. In the administration interface, go to **PBX Services**.
2. Double-click **Call monitoring**.
3. In the **Edit Service** dialog, you can change the service extension.
4. Check the **Service is enabled** option.
5. Read the disclaimer carefully and click **I Agree**.



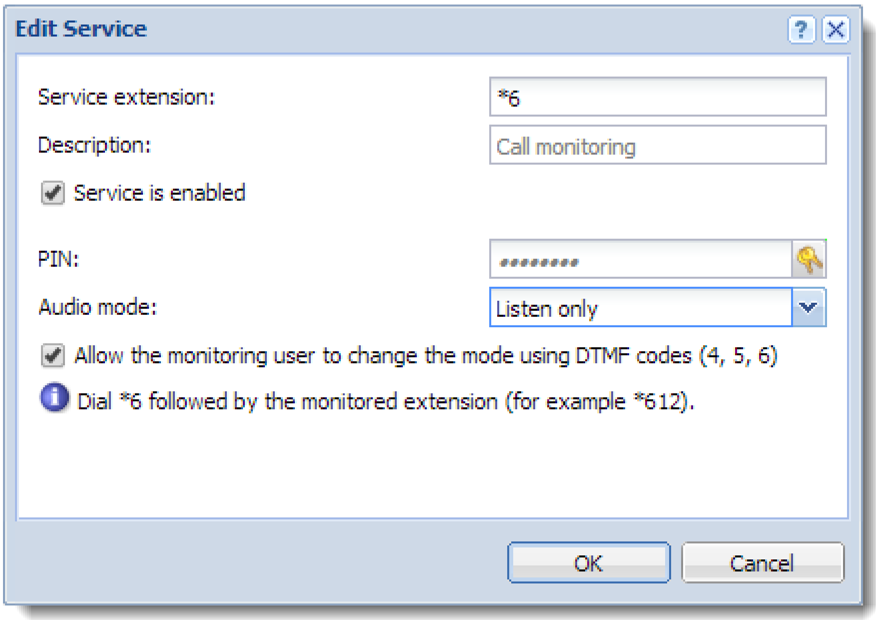
6. Click the keys icon and remember the PIN number. You can also change the PIN number. The PIN protects the call monitoring from misuse.

7. Select the **Audio mode**:

- Listen only — muted: When joining an active call in listen only mode, there is no indication to the active callers that you have joined the call.
- Whisper to the extension only — muted only to remote party
- Talk to both — unmuted

8. To allow users to change the audio mode with DTMF codes, check the **Allow the monitoring user to change the mode using DTMF codes (4, 5, 6)** option. Users can change the audio mode with a key on their phone devices (**4** is for **Listen only**, **5** is for **Whisper to the extension only** and **6** is for **Talk to both**).

9. Click **OK**



The call monitoring service is configured.

Setting call permissions

Set a call permission group for users who can use the call monitoring feature (people who knows the PIN number):

- call monitoring is allowed on extensions, which can be monitored (rules 1, 2 and 3 in the figure)
- other calls with *6 are forbidden (rule 4 in the figure)

Example:

The first three rules allow call monitoring on extensions 111, 112, 113:

1. In the administration interface, go to **Definitions > Call Permission Groups**.

2. Click **Add**.

3. In the **Add Call Permission Group** dialog, add the name of the group.

4. In the **Description** field, type `Group restricts call monitoring to listed extensions`.

5. Click **Add**.

6. In the **Add Prefix** dialog, type *6111.

7. Switch the rule to **Allowed** and click **OK**

8. Repeat the steps 5, 6 and 7 for extensions 112 and 113.

The fourth rule disables general usage of *6 prefix:

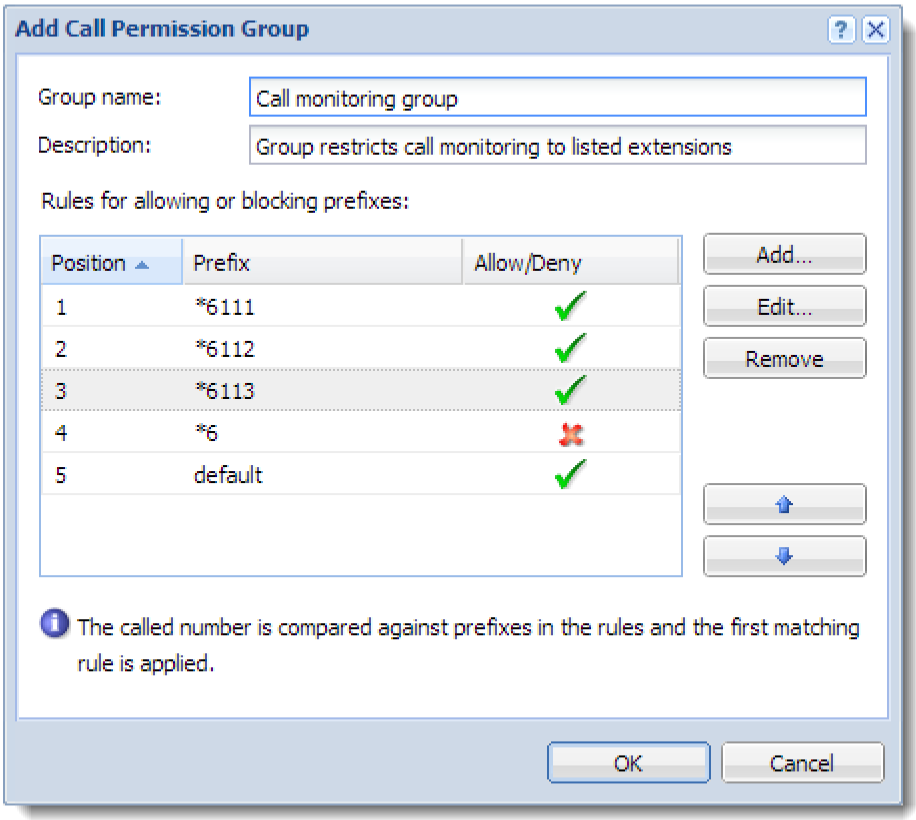
1. Click **Add** in the **Add Call Permission Group** dialog (it is still opened).

2. In the **Add Prefix** dialog, type *6.

3. Switch the rule to **Denied** and click **OK**Now, you can compare your result with figure. They should be the same.

> **WARNING**
> The denial rule must be placed below the allowing rules.

4. Click **OK** in the **Add Call Permission Group** dialog.



The group for call monitoring is established.

Now, you must assign the group to users eligible to use the call monitoring prefix and know the PIN number:

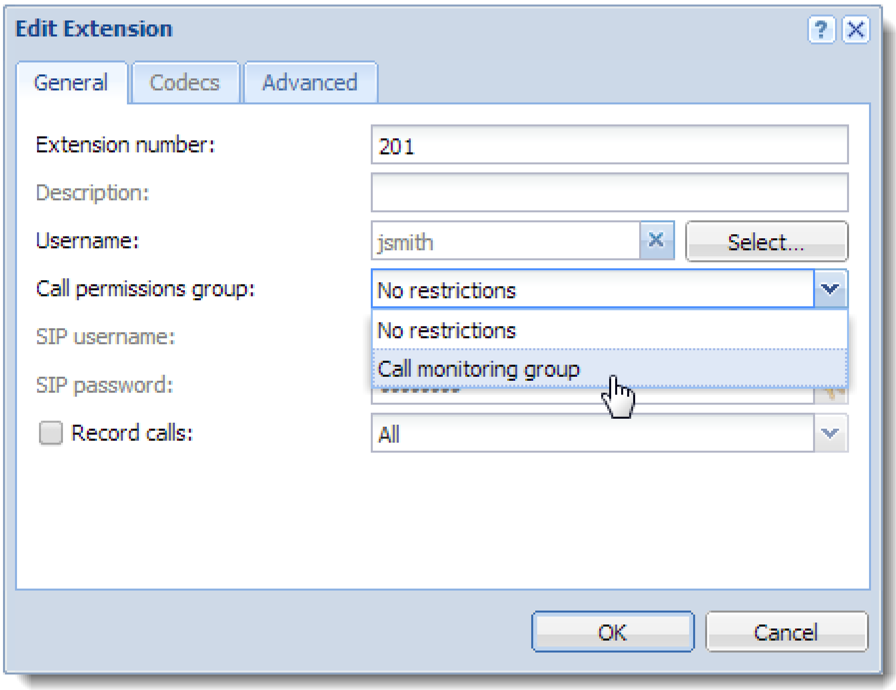
1. In the administration interface, go to **Configuration > Extensions**.

2. Select an extension assigned to John Smith (in figure it is extension 201) and click the **Edit** button.

3. In the **Edit Extension** dialog, change **Call permissions group** to **Call monitoring group** (see screenshot).

4. Click OK.

5. If the user has assigned more extensions, you must set **Call monitoring group** for all of them to avoid a risk of misuse of the call monitoring.



The cal monitoring group is assigned the user who is eligible to use the call monitoring prefix.

Using call monitoring

To use the call monitoring service you must know:

- the service extension (***6** by default),
- the PIN,
- the monitored extension (for example **111**).

For extension **111**, dial ***6111** to listen to the conversation. Then, you will be asked for the PIN number. Now, you are silently connected to the call on extension **111**.

If you are connected to the **111** extension, you can change a mode during the call (if allowed by the call monitoring service):

- press **4** for listen only mode
- press **5** for whisper to the extension only
- press **6** for talk to both

You can also monitor all employees in your office:

- extensions in your office start with 11
- five of them are assigned to employees (111, 112, 113, 114, 115)

If you dial ***611**, you can connect to the first ongoing call from all extensions starting with 11

If you dial ***61**, you can connect to the first ongoing call from all extensions starting with 1

If you dial ***6**, you can connect to the first ongoing call from all extensions of your telephony subsystem.

Pressing * key will look for another call to monitor.

> **WARNING**
> As you can see, the user can monitor all calls in your telephony subsystem. Therefore, it is important to set call permissions for all users, who are eligible to use the call monitoring prefix.

4 Settings

This section contains information about:

4.1 Phone provisioning

This section contains information about:

4.1.1 Configuring automatic phone provisioning

> **NOTE**
> Watch the Configuring automatic phone provisioning in Kerio Operator video.

Phone provisioning is used for automatic configurations of selected hardware SIP phones. Phone provisioning means:

- phone automatically connects to the PBX after booting and is assigned a phone extension,
- extensions are managed in the administration interface,
- if you confirm or plan it, the system will perform an automatic restart of provisioned phones if needed,
- phone firmware is automatically updated,

- displaying a company logo on hardware phones supported by Kerio Operator
- accessing company contacts through LDAP

NOTE

Automatic firmware update is not supported for the Polycom phones and the original Cisco phones (Cisco SPA is supported). However, there is a possibility to update the firmware. You can upload all necessary files to folder `/var/tftp` in Kerio Operator manually. For more information, refer to Uploading configuration files to Kerio Operator TFTP server (page 190).

IMPORTANT

Use of phone provisioning is not always suitable. If Kerio Operator is located and runs in the Internet, for security reasons we do not recommend to use automatic phone provisioning.

What you need

1. In your local network, you need a DHCP server supporting parameter 66 (TFTP server address). Enter the address of Kerio Operator in this parameter.

NOTE

For more information, refer to Configuring parameter 66 in DHCP server in Kerio Control (page 269).

2. Only selected phones support automatic phone provisioning.

3. Appropriate settings need to be done in Kerio Operator.

NOTE

If you wish to connect a phone which is not currently supported in Kerio Operator, you cannot use automatic provisioning. The configuration must be done on the hardware phone.

How to add a phone

1. In the administration interface, go to **Provisioned Phones > Hardware Phones**.

2. Click **Provisioning Settings**. The configuration dialog windows is opened.

3. Check the **Enable provisioning** option. The option must be checked.

4. Check option **Create new extension for newly registered phones** in case you create users locally (do not map them from a directory service).

WARNING

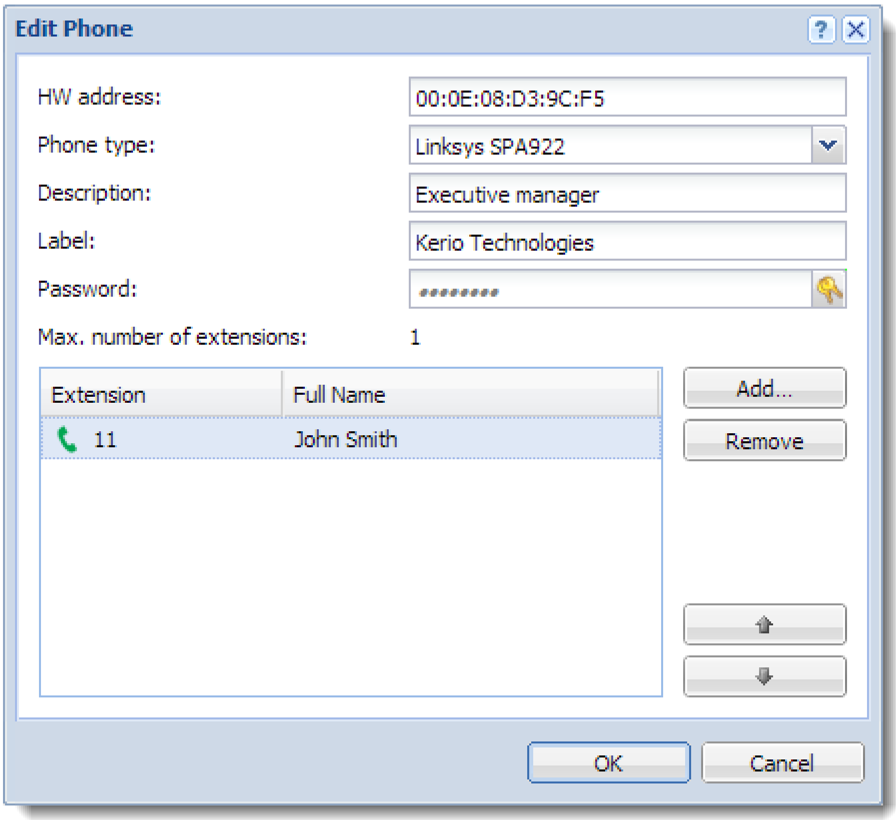
The **Create new extension for newly registered phones** option is checked by default. If you uncheck it, you cannot use automatic remote phone restart — you will have to restart phones manually if needed.

5. Each telephone must be authenticated when connecting to the PBX. Extension number and password are used for SIP authentication (Master Password in this case). Option **Master password for phones is enabled** enables to create one password for all provisioned phones. The password is saved in the configuration file which is sent to the phone upon the first connection to the network and the phone will use this password to authenticate at Kerio Operator. If you disable option **Master password for phones is enabled**, all phones will have their own passwords (it can be viewed in the configuration dialog of each phone).

Now the general environment for the provisioned phones is configured. Once a phone is connected to your network, it will be listed in section **Provisioned Phones**.

Adding phones manually

Phones which are not connected to the network can also be provisioned. You may do so manually — you need the phone's hardware address and the type of the phone. The procedure is described below:



Screenshot 29: Connecting a phone manually

1. In section **Provisioned Phones**, click **Add**.

2. This opens a dialog which requires the hardware address of the phone (MAC address of the network card in the phone). The address may lack the colons. Once you save it, the colons will be added automatically.

3. Select the correct type of the hardware phone (special configuration scripts are created according to the phone type).

4. (Optional) Set a label of the phone (for example the name of your company). The upper label on the phone display.

5. Assign the phone user or users who will use it.

> **NOTE**
> If you do not know to which person the extension will be assigned, check option **Generate new extension number** and the extension will be assigned automatically. Phones without extensions assigned cannot be provisioned.

Importing from CSV file

Phones can be imported from a CSV file. Data in the file must follow certain rules:

- hwAddress — hardware address of the phone,
- phoneManufacturer — name of the phone's manufacturer,
- phoneType — phone type,
- extension1; extension2; ... — extensions assigned to the phone. The maximum number of extensions depends on the phone type.

Each phone uses one line and all items are separated by a semicolon.

The file may look as follows:

```
00:1a:a0:be:1e:cd;Cisco;7940;111;112
00:1b:b0:cd:e1:ca;Cisco;7960;115
00:1c:c0:ab:a2:24;Linksys;SPA942;113;114
```

Import data from a CSV file as described below:

1. In the **Provisioned Phones** section, click on **Advanced > Import from a CSV file**.

2. This opens dialog **Import from a CSV file** — click on **Upload CSV file**.

3. If the data in the file are correct, a list of all the phones and extensions is displayed. Check those you want to import.

4. Click **OK**

5. The imported phones are displayed in the **Provisioned Phones** table.

Restarting provisioned phones

When you change configuration which affects provisioned phones, the phones need to be restarted (for example, when you create a new call route). When you do so, a dialog window recommending phone restart is displayed. You can do it immediately or wait for a more convenient time (for example to an off-peak time). To restart phones later:

1. Open the **Provisioned Phones** section.

2. Click **Advanced > Restart All Phones**.

> **WARNING**
> Some Cisco telephones from newer series are not able to restart automatically. In case of configuration changes you have to check the result. If anything is wrong, restart the phones manually.
> This warning doesn't relate to Cisco SPA phones.

Firmware

Kerio Operator allows easy installation of phone firmware which are managed through the phone provisioning:

1. Go to section **Provisioned Phones** and click on the **Advanced > Firmwares** button.

2. In the **Firmwares and Logos** dialog, select a firmware and click **Edit**.

3. In the **Edit firmware** dialog, select Verify the firmware. Kerio Operator vrifies if the firmware includes all important files and information.

4. Click **Upload File**.

5. This opens a dialog where you select a firmware file and confirm the selection.

6. In the **New firmware** dialog, select the appropriate phone.

7. Click **OK**

The new firmware is installed and after the restart will be installed to phones.

Uploading a phone provisioning module

If you want to change or create a provisioning module (archived templates + PHP scripts which can change phones behavior), download Provisioning Developer Documentation and read it carefully.

When the provisioning module is prepared and archived, upload it to Kerio Operator:

1. Go to administration interface.
2. In section **Provisioned Phones**, click **Advanced > Provisioning Modules.**.
3. Click **Upload**.
4. Restart your phones.

Overriding templates

For more information, refer to Editing provisioning templates (page 180).

What to do if you want to know the password of your phone

If any of your users needs to know the password of their phone, we do not recommend to provide them with the Master Password. We have a specific solution:

1. In the administration interface, go to **Provisioned Phones > Hardware Phones**.
2. Click **Provisioning Settings**.
3. Disable master password.

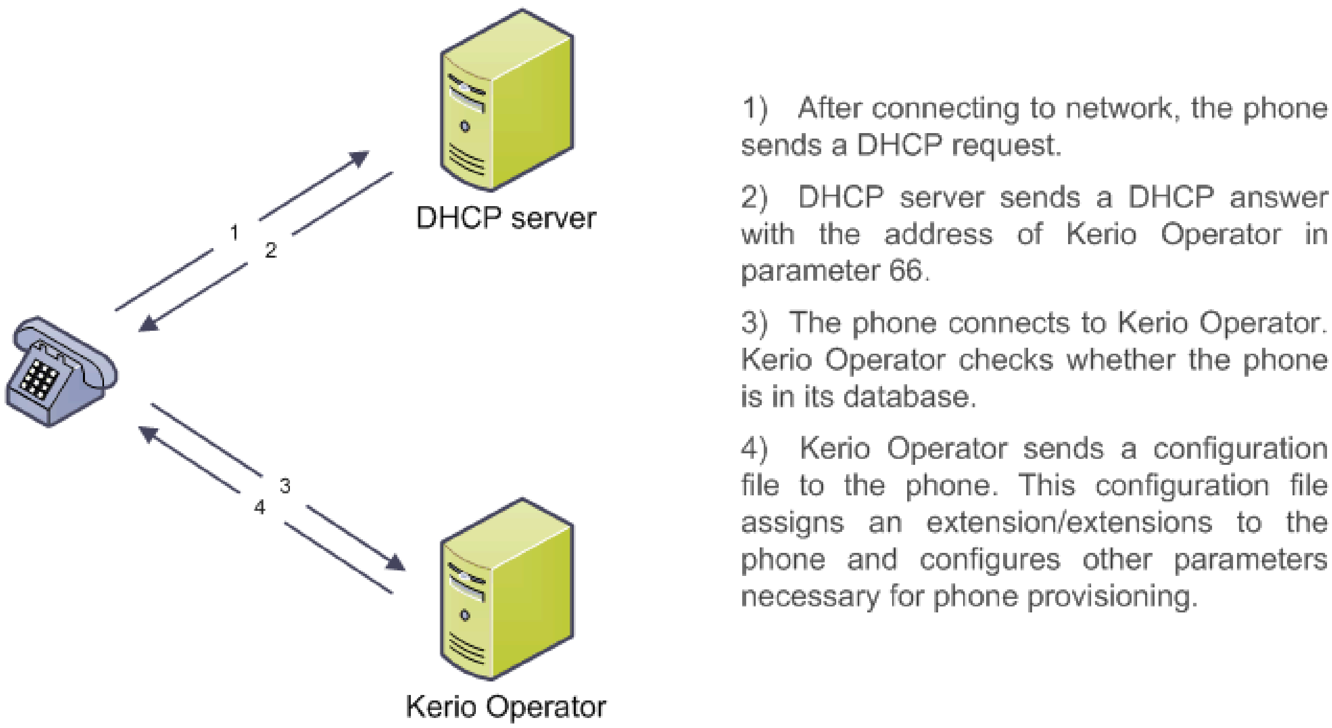
Once you disable it, each phone will have their own password which can be shared with individual users.

Configuring inter-digit timeout

Inter-digit timeout sets the time between dialing the last digit and automatic dial. If your users complains that it is too long or too short, you can adjust it:

1. Go to the administration interface.
2. In section **Provisioned Phones**, go to **Provisioning Settings**.
3. In the **Phone Provisioning Settings**, set the **Inter-digit timeout**.

How phone provisioning works



Screenshot 30: Automatic HW phone provisioning

This is how the automatic phone provisioning works:

» The telephone boots in the network and sends a DHCP request for an IP address.

» DHCP server accepts the request, assigns an IP address and sends it back in a DHCP reply. Besides the IP address, the message also contains TFTP (Trivial File Transfer Protocol) server address — Kerio Operator, in our case.

» SIP phone connects to TFTP server integrated in Kerio Operator.

» Kerio Operator checks whether the phone is new:

- if it is new, Kerio Operator generates a new phone extension for the phone;
- if it is not new, Kerio Operator finds the extension which the phone has used.

» Kerio Operator generates a configuration file suitable for the particular phone type and sends it via the TFTP protocol.

» The phone is configured using the values it has acquired in the configuration file and is ready to be used.

> **NOTE**
> Some phones perform an automatic restart during the configuration.

4.1.2 Provisioning of Kerio Operator Softphone for mobile devices

This topic describes auto-provisioning of Kerio Operator Softphone for mobile devices. Auto-provisioning and its functionality for SIP phones is described in Configuring automatic phone provisioning.

Prerequisites

» Kerio Operator must have a DNS name. Type the DNS name in the **Configuration > Network** section.

WARNING

To secure your Kerio Operator Softphones on Android devices, you must have a fully qualified domain name in the SSL certificate of the Kerio Operator server.

» Kerio Operator must use a valid SSL certificate. The certificate name must correspond with the Kerio Operator DNS name. For more information, see Securing Kerio Operator Softphone with SSL certificates.

Configuring provisioning for Kerio Operator Softphone

Step 1	Add a new extension or a new registration of their existing extension to users who want to use Kerio Operator Softphone.
Step 2	Add users to provisioning: 1. In the administration interface, go to **Configuration > Provisioned Phones > Softphones** 2. Click **Add**. 3. In the **Select User** dialog, select the user who wants to use Kerio Operator Softphone. 4. Save the settings.
Step 3	Users must configure their mobile devices to connect to Kerio Operator.

Securing Kerio Operator Softphone with SSL certificates

To secure your Kerio Operator Softphones, you must have one of the following SSL certificates:

» A paid SSL certificate signed by a certification authority. These certificates do not require any further configuration.

WARNING

Do not use wildcard certificates. Kerio Operator Softphone follows the RFC 5922 standard.

» A self-signed certificate created by your Kerio Operator server. If you use a self-signed certificate, users must download and install the certificate manually. For more information, go to Using the self-signed certificate from your Kerio Operator server.

Configuring a dial plan

Users with Kerio Operator Softphone want to use their contact list, where phone numbers are stored in different formats. The Dial Plan translates phone numbers from the format used in a user's contact lists to the format that can be dialed via your Kerio Operator PBX:

1. In the administration interface, go to **Provisioned Phones > Softphones**.
2. Click **Dial Plan Configuration**.
3. Click **Add** to create a new rule.
4. Save the rule and click **Test** in the **Dial Plan Configuration** dialog.
5. If you need more rules, create another one.
6. Sort rules from specific to general. Rules are applied from top to bottom.
7. Save the settings.

Creating rules

You can use the following characters when creating new rules.

Character	Description
0 to 9	digits
x	a single wildcard
*#+	Keyboard symbols
[ ]	A collection that can include a range. For example [6-9] means 6 7 8 9. Or [136-9] means 1 3 6 7 8 9.
.	Repeat the last element 0 or more times. For example, with the pattern 12. the following input will match: 1 (The 2 is repeated zero times) 12, 122, 1222 and so on

Screenshot 31: Characters for your dial plan

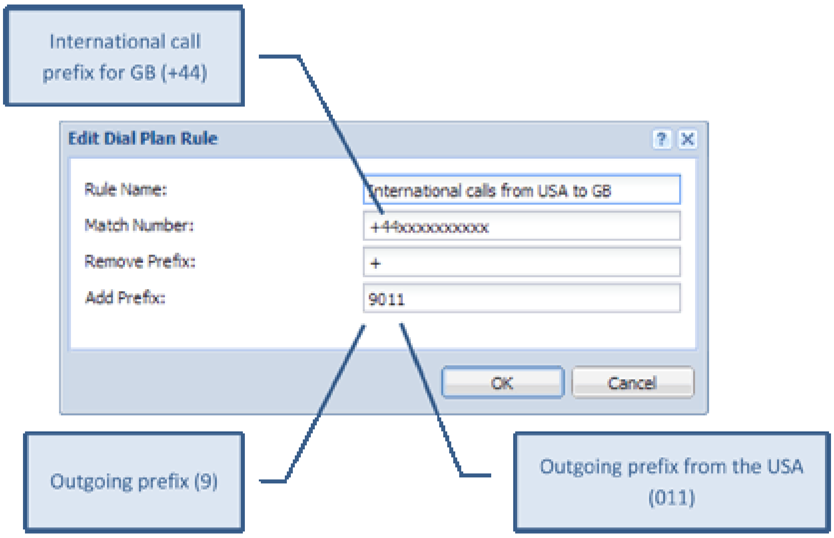
Example 1: International calls from USA

Match number: `+x.`

Remove prefix: +

Add prefix: `011`

The following image describes a scenario when you wan to call from the USA (prefix `011`) to GB (prefix `+44`) and outgoing prefix of your company is 9.



Screenshot 32: International calls from USA to GB

Example 2: Outgoing prefix 9

Match number: `x.`

Remove prefix: leave empty.

Add prefix: 9

Example 3: International calls in Europe (replacing + by 00)

Match number: `+x.`

Remove prefix: +

Add prefix: `00`

4.1.3 Accessing company contacts through LDAP on provisioned phones

Kerio Operator offers searching in your LDAP directory from your provisioned phones.

WARNING

Cisco79xx phones are not supported.

Polycom phones are not supported with Kerio Connect LDAP.

Connecting to Kerio Connect LDAP/Microsoft Active Directory

1. In the administration interface, go to **Provisioned Phones**.

2. Click the **Provisioning Settings** button.

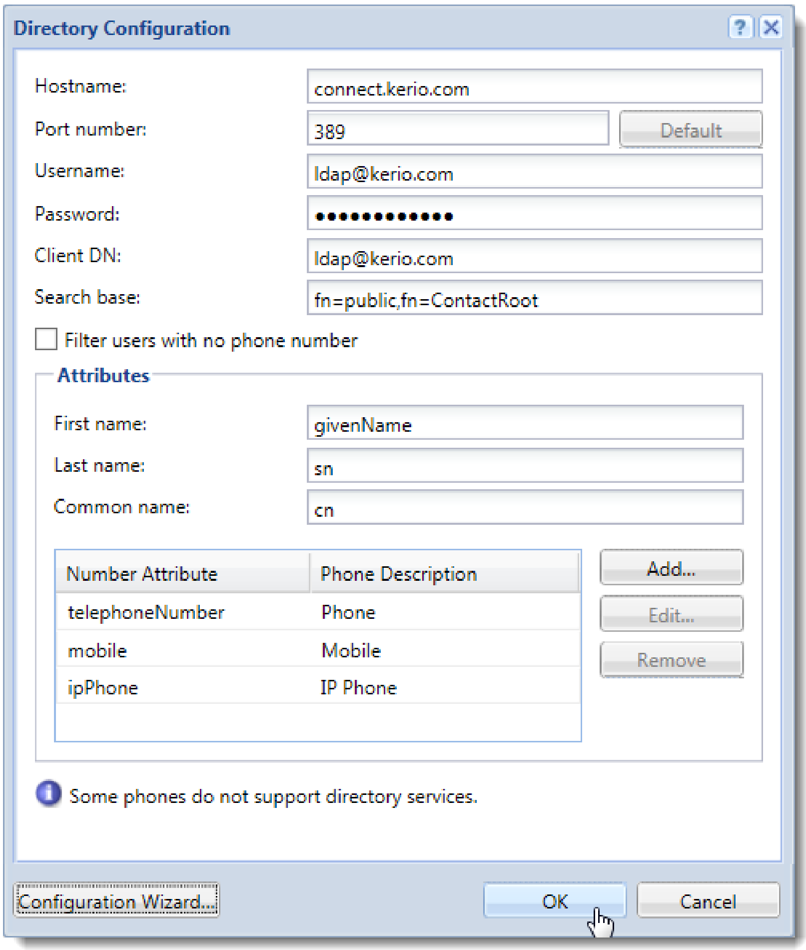
3. In the **Phone Provisioning Settings** dialog, select option **Directory configuration is enabled**.

4. Click **Configure**.

5. Click **Configuration Wizard**.

6. Select type of a service:

- Kerio Connect LDAP — type Kerio Connect hostname, username and password.
- Active Directory — type domain name and hostname of your Active Directory and credentials of account with at least read-only access to Active Directory,



Screenshot 33: The Directory Configuration dialog after finishing Kerio Connect LDAP configuration

NOTE

We recommend to create a special account with read-only access and use credentials of this account.

3. Save the settings.

4. In **Provisioned Phones**, click **Advanced** and restart all provisioned phones. Phones read the new configuration and start to communicate directly with the LDAP server.

Try this feature on your phone. Find a directory on the phone and check the contact list.

For information on how to use your phone directory, read the manual of your phone.

Connecting to LDAP in general

1. In the administration interface, go to **Provisioned Phones**.

2. Click the **Provisioning Settings** button.

3. In the **Phone Provisioning Settings** dialog, select option **Directory configuration is enabled**.

4. Click **Configure**.

5. Fill the **Directory Configuration** dialog.

6. Save the settings.

7. In **Provisioned Phones**, click **Advanced** and restart all provisioned phones. Phones read the new configuration and start to communicate directly with LDAP server.

Try this feature on your phone. Find a directory on the phone and check the contact list.

For information on how to use your phone directory, read the manual of your phone.

4.1.4 Using provisioning tools

> **NOTE**
> New in Kerio Operator 2.3!

Kerio Operator includes tools for phone administration. These tools can:

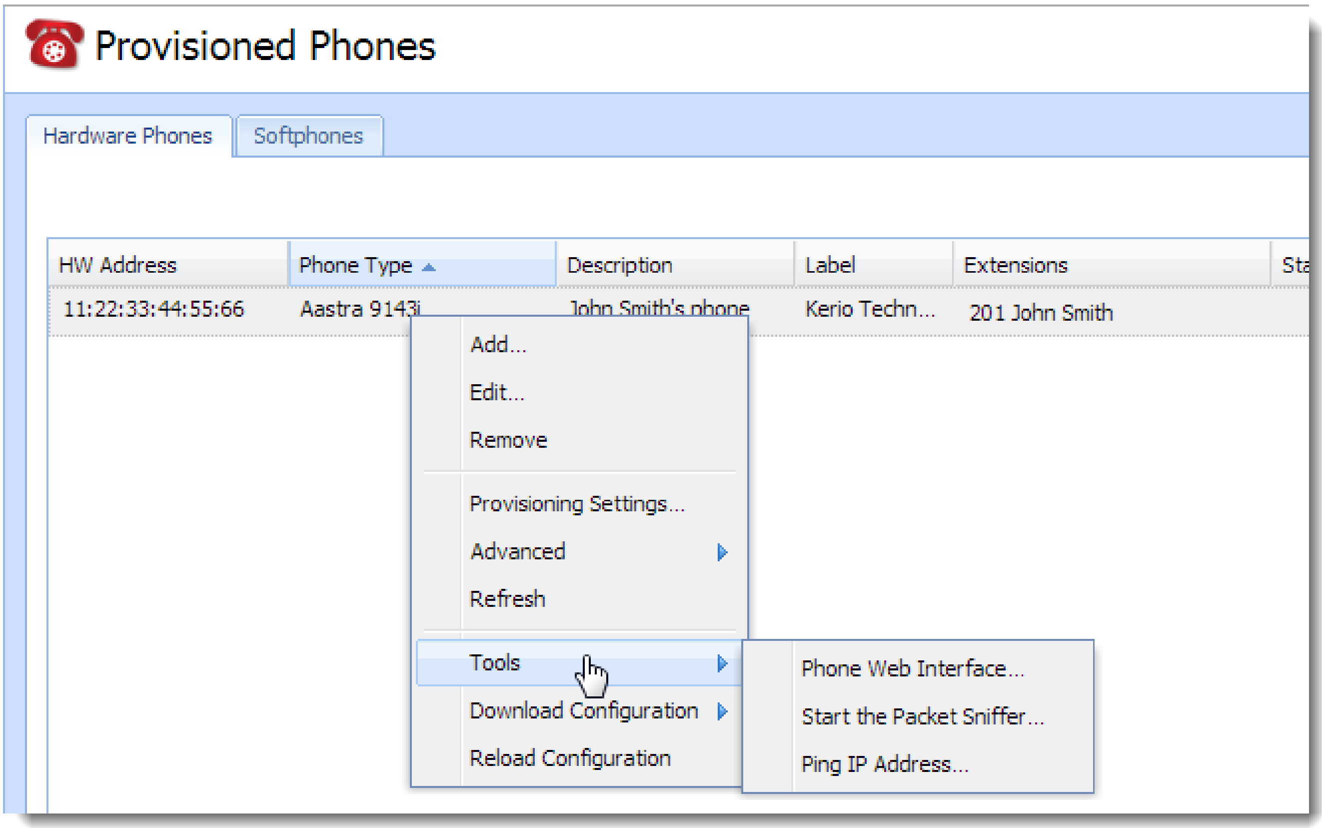
- display the phone web interface.
- open a packet sniffer for a communication between the phone and Kerio Operator.
- ping IP address of the phone.

Using provisioning tools

1. In the administration interface, go to **Provisioned Phones**.

2. Right-click a provisioned phone and in the context menu select **Tools**.

3. Select a tool and use it.



4.1.5 Editing provisioning templates

Kerio Operator offers overriding templates for provisioned phones.

Template overrides were developed to allow you to change default values in the provisioning template. You can change for example remote changing BLF, speed dials or phone's language).

Using Template Overrides

1. Go to administration interface.
2. In section **Provisioned Phones**, click **Advanced > Template Overrides.**.
3. Click **Edit**.
4. Edit the template. Kerio Operator uses the same file for a group of phones, for example for all snom phones or all Cisco SPA phones.
5. Click **OK**
6. Click **Close**.
7. Restart your phones.

Using developer's documentation

When you edit a template, you can use input variables that come preset when your scripts are run. Variables are described in **Kerio Operator provisioning reference guide**:

1. Download Provisioning Developer Documentation.
2. Extract the ZIP file.

3. Open **refguide.pdf**.

You can use all variables mentioned in the downloaded document.

Example: Changing a phone's language

Lines beginning with `@` contain regular PHP code which is executed when the template is sent to the phone. So you can do things like:

```
@ if ($IDENT === 'snom720') {
language: Deutsch
@ }
```

This switches the snom 720 phones' language to German.

To find out what the variable `$IDENT` can contain, you can temporary add line:

```
@ var_dump($PHONE_TYPES);
```

To view the result:

1. In the **Provisioned Phones > Hardware Phones** section, right-click the snom phone.

2. In the context menu, click **Download Configuration** and select the correct interface.

3. Open the saved archive and verify the variable in the file.

To differentiate between the phones, you can use variables such as `$PHONE_IP` or the `$LINES` array. To change a specific phone, one can do:

```
@ if ($PHONE_IP === '192.168.12.11') {
language: Deutsch
@ }
```

> **NOTE**
> In this case you can not verify the functionality by downloading the file in the administration, because the `$PHONE_IP` does not match.

Another example is:

```
@ if (isset($LINES[0]) && $LINES[0]['TELNUM'] === '21') {
language: Deutsch
@ }
```

The example checks if the phone has at least one extension number and sets the language to German if the first extension is `21`.

4.1.6 Displaying your company logo on the provisioned phones

You can display your company logo on hardware phones supported by Kerio Operator.

What you need

- Logo — each phone firmware needs a logo in a different format.
- Phones must be provisioned.

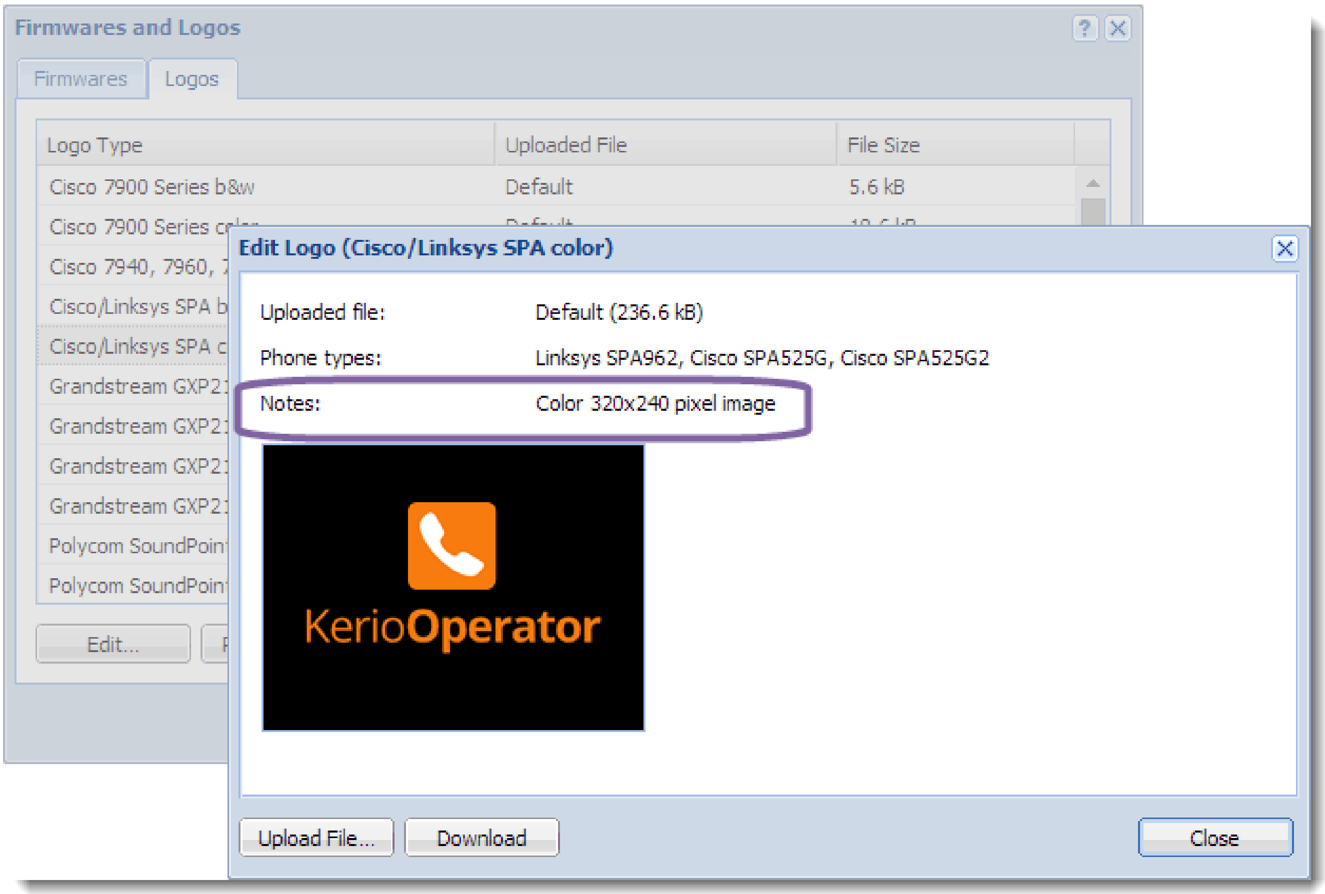
Which type of logo do you need

1. In the administration interface, go to **Provisioned Phones**.

2. Click the **Advanced > Logos** button.

3. In the **Firmwares and Logos** dialog, go to tab **Logos**.

4. Find the firmware type installed on your phones and click **Edit**. In **Notes**, you can find the logo parameters.



Screenshot 34: Logo parameters in the Edit Logo dialog

Adding your logo to phones

The Kerio Operator logo is set by default and you have to change it:

1. In the administration interface, go to **Provisioned Phones**.

2. Click the **Advanced > Logos** button.

3. Find the logo type for your phone and click **Edit**.

4. Click **Upload File** and upload your logo.

5. Close the dialog.

6. In **Provisioned Phones**, click the **Provisioning Settings** button.

7. In the **Phone Provisioning Settings** dialog, select **Display logo on the screen**.

8. Save the settings.

9. Restart all phones manually.

4.1.7 How to configure phone provisioning on Polycom phones

This topic takes you through the configuration of the automatic phone provisioning via TFTP on Polycom phones.

1. After the phone boots, click the **Menu** button.

2. Select **Settings** and confirm.

3. Select **Advanced** and confirm.

4. Now the phone requires a password for phone provisioning.

5. Enter the default password which is 456 (once the provisioning is configured, the password will be synchronized with the one set in the Kerio Operator administration interface in section **Provisioned Phones**).

6. Select **Admin Settings** and confirm.

7. Select **Network Configuration** and confirm.

8. Select **Server Menu** and confirm

9. Select **Server Type** and use the right arrow to switch to **Trivial FTP**.

10. Press **Exit** to exit the **Settings**.

4.1.8 Phone provisioning - wrong detection of CISCO phones

Problem

- You cannot configure a provisioned phone.
- The phone has be detected and the administration interface in section **Provisioned Phones** displays a different type of phone.
- The phone does not accept the extension it has been assigned.

Explanation

Some phones support two types of protocols:

- SCCP
- SIP

The default protocol is set to SCCP. Phones may not detect the correct communication protocol. As Kerio Operator uses SIP for communication, the telephone cannot be attended.

How to solve it?

Switch the communication protocol of the phone from SCCP to SIP manually.

Example: Cisco SPA 525

1. Go to the phone configuration interface in your browser: `https://phoneIPaddress/admin/advanced`

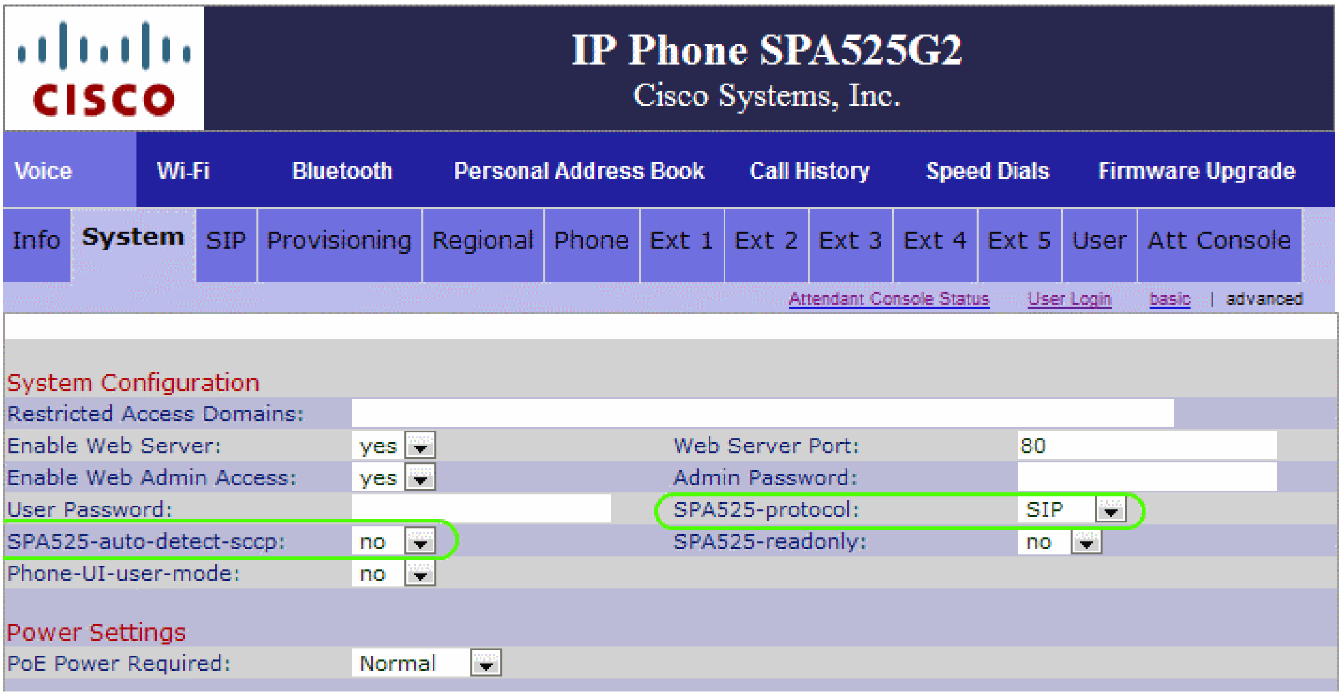
2. Change **SPA525-protocol** to **SIP**.

3. Change **SPA525-auto-detect-sccp** to **no**.

4. Login to Kerio Operator and go to **Provisioned Phones**.

5. Remove the wrongly detected phone (it has been detected as an older CISCO phone).

6. Restart the phone (so called soft restart).



Screenshot 35: Configuring the Cisco SPA 525 phone

> **NOTE**
> A similar problem has been identified for CISCO SPA 303. In this case, the phone's administration is locked (we have found the solution on https://supportforums.cisco.com). Further steps are similar to Cisco SPA 525.

4.1.9 Uploading configuration files to Kerio Operator TFTP server

Why to use phone or other device configuration file

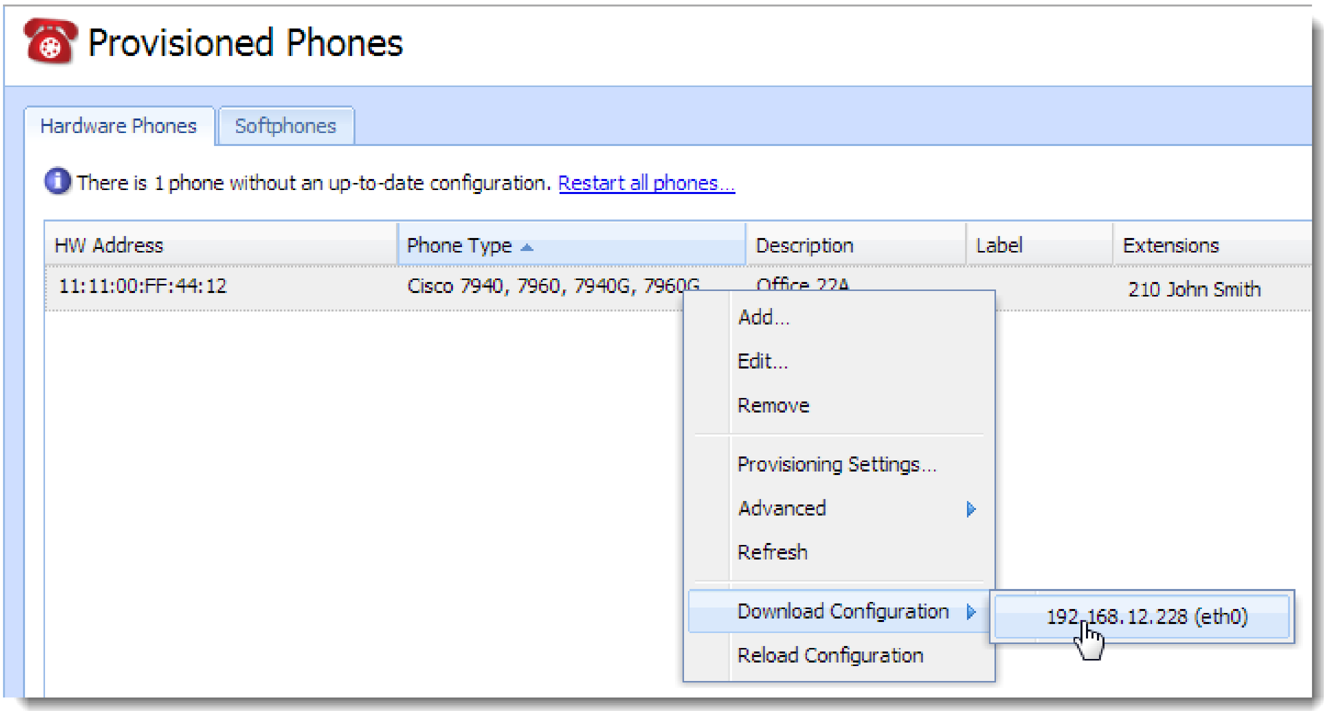
- phone provisioning of unsupported devices (hardware phones or other devices with a TFTP client)
- phone firmware upgrade
- BLF configuration, ring tones (different ring tones for different phones)
- password change for all extension assigned to one phone

Obtaining the configuration file

The following instructions will come in handy, if you wish to change the configuration file of a provisioned phone:

1. In the administration interface, go to **Configuration > Provisioned Phones**.

2. Right-click the phone whose configuration file you wish to download.



Screenshot 36: Downloading the configuration

3. Click **Download Configuration** and select the interface. Each interface has a different configuration — different IP addresses.

4. The ZIP file with the current configuration will be automatically saved on your computer.

Uploading new or changed configuration files to Kerio Operator

What you need

The file must be uploaded via SSH using SCP.

Locate configuration files to `/var/tftp`

How to enable SSH in Kerio Operator

Follow these instructions:

1. In the administration interface, go to section **Status > System Health**.

2. Click **Tasks** while pressing the **Shift** key.

3. Select **Enable SSH**.

4. Connect to Kerio Operator via SCP (use for example WinSCP for Windows) and upload the file via SSH using SCP. For access use username `root` and password of a Kerio Operator administrator.

4.2 Accounts

This section helps you create user accounts and their phone extensions.

4.2.1 Creating user accounts

User accounts in Kerio Operator are used for:

- Login users to Kerio Phone
- Link users with an extension
- Set access rights to the system

Adding new accounts

You can create either a local user or map existing users from a directory service.

Adding local accounts

If you do not use directory services, create a local user in the Kerio Operator administration:

1. In the **Configuration > Users** section, click **Add**.
2. The **Add User** dialog box opens.
3. On the **General** tab, type username and password. The username must not contain spaces, diacritics and special symbols.
4. Click OK.

The user account appears in the **Users** section and the user can connect to Phone.

Adding accounts from directory service

Mapping differs according to the directory service used:

- Microsoft Active Directory
- Apple Open Directory

You need basic login credentials to connect directory service to Kerio Operator.

For more information, refer to Connecting Kerio Operator to directory service (page 275).

Assigning extensions to users

An extension is an internal telephone line. Each user can have assigned one or more extensions in Kerio Operator.

1. In the **Configuration > Users** section, select a user and click **Edit**. The **Edit User** dialog box opens.
2. On the **Extensions** tab, click **Add**. The **Select Extensions** dialog box opens.
3. In the **Select Extensions** dialog box, click **Add**. The **Add Extension** dialog box opens with predefined unused extension.
4. If the extension number meets your dial plan, click OK. If not, rewrite the extension number and then click OK.
5. Save the settings.

The users can use their Kerio Operator phone extension.

For more information, refer to Creating extensions (page 193).

Configuring ringing rules

For more information, refer to Redirecting calls (page 209).

4.2.2 Creating extensions

An extension is an internal telephone line. Each user can have assigned one or more extensions in Kerio Operator.

The total number of extensions is limited to three times the number of licensed users.

> **NOTE**
> Service extensions configured on the PBX services tab are not counted by the license file.

Adding new extensions

You have three options to add a new extension:

- An extension is created automatically when you connect a provisioned phone to the network.
- You can create an extension in **Configuration > Users** — the extension is assigned to a particular user.
- Create an extension in **Configuration > Extensions** — the extension is created as standalone (without being assigned to a user).

Creating a standalone extension

If you have a phone which is not used by any particular user, you can create a standalone extension for it.

1. In the administration interface, go to **Configuration > Extensions**.

2. Click **Add > Add Extension**.

3. Type an extension number. The field suggests an unused extension. You can change the extension number manually if necessary.

4. Save the settings.

SIP username and SIP password

Each extension has a SIP username and a SIP password. Kerio Operator uses SIP usernames and SIP passwords for authentication of phones to Kerio Operator. You use SIP username/password for connecting softphones or hardware phones to Kerio Operator. For more information, refer to Configuring multiple registration of an extension (page 194).

SIP usernames/passwords cannot be used to login into Kerio Operator or Kerio Phone.

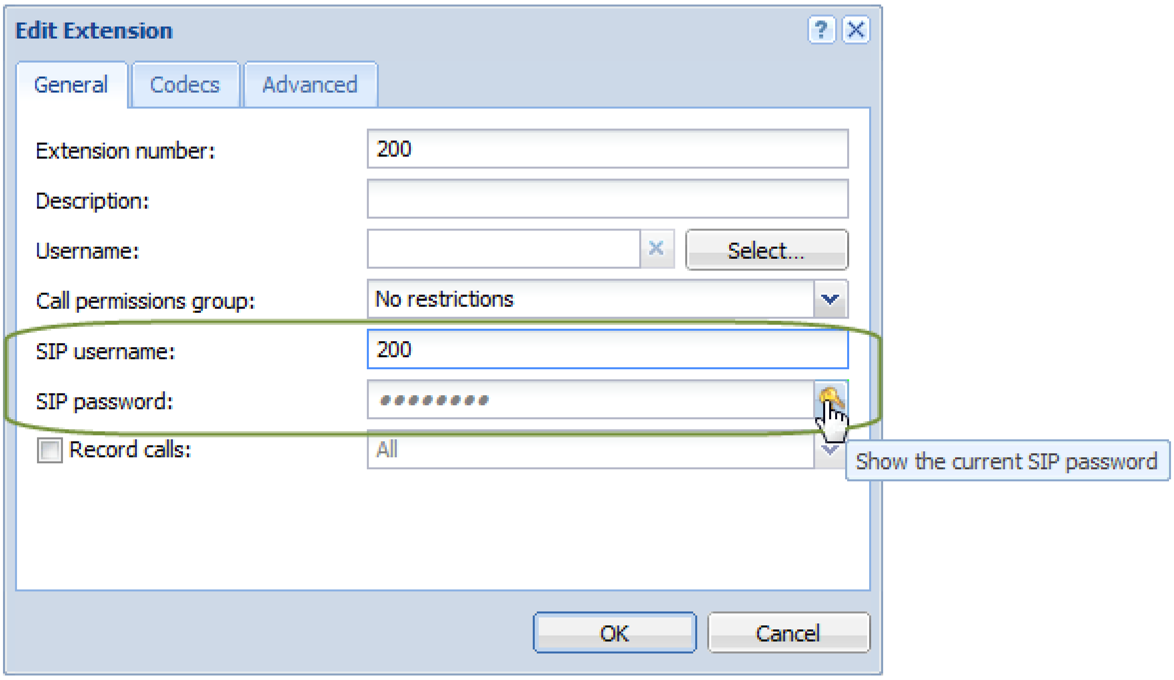
Using SIP username/password

1. In the Kerio Operator administration interface, go to **Configuration > Extensions**.

2. Select an extension and click **Edit...**.

3. In the **Edit Extension** dialog, you can see fields **SIP username** and **SIP password**.

4. To display the SIP password, click the keys icon.



Screenshot 37: SIP username and SIP password

Now you can view the SIP username/password and use it for connecting a phone to Kerio Operator.

Encrypting calls

In Kerio Operator, you can encrypt your calls for any extensions.

1. In the Kerio Operator administration interface, go to **Configuration > Extensions**.

2. Select an extension and click **Edit...**

3. Click the **Advanced** tab and select **Encrypt communication (TLS and SRTP)**.

4. Click **OK**

Now Kerio Operator encrypts all calls for the selected extension.

For more information, refer to Securing Kerio Operator (page 254).

4.2.3 Configuring multiple registration of an extension

Do you want to use your extension with various phones? Softphone in your cell phone or IP phone in your smartphone? The solution is multiple registration.

> **NOTE**
> Multiple registration (in contrary to assigning more extensions to one user) gives user the possibility to call from the same extension any time they make a call.

EXAMPLE

User Brenda Roar with username `broar` working at the Marketing department uses the extension 224. When necessary, she also works from home. She uses the following to communicate:

1. She has an automatically provisioned phone Cisco 7940 in his office.

2. She has X-Lite softphone on her home computer.

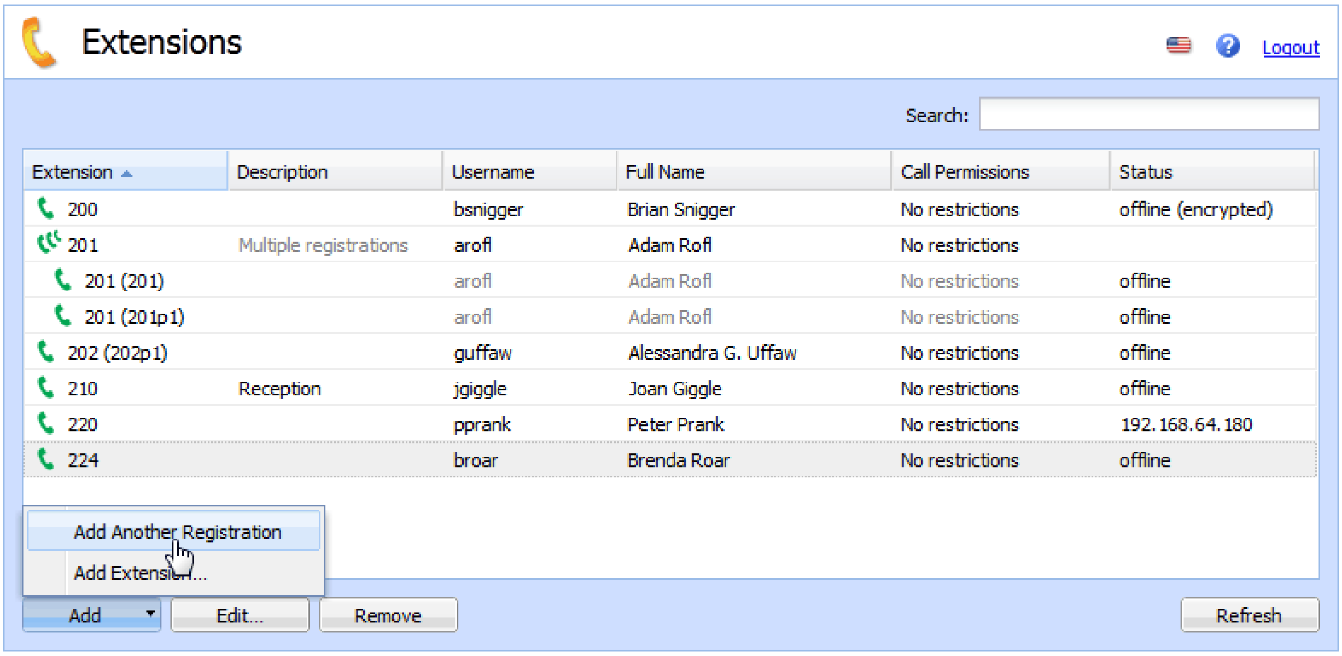
3. Occasionally, when connected via WiFi, she uses a SIP client on her mobile phone.

With correct settings of multiple registration that will be described in the following chapter she can use all the before-mentioned methods to authenticate.

Creating multiple registrations

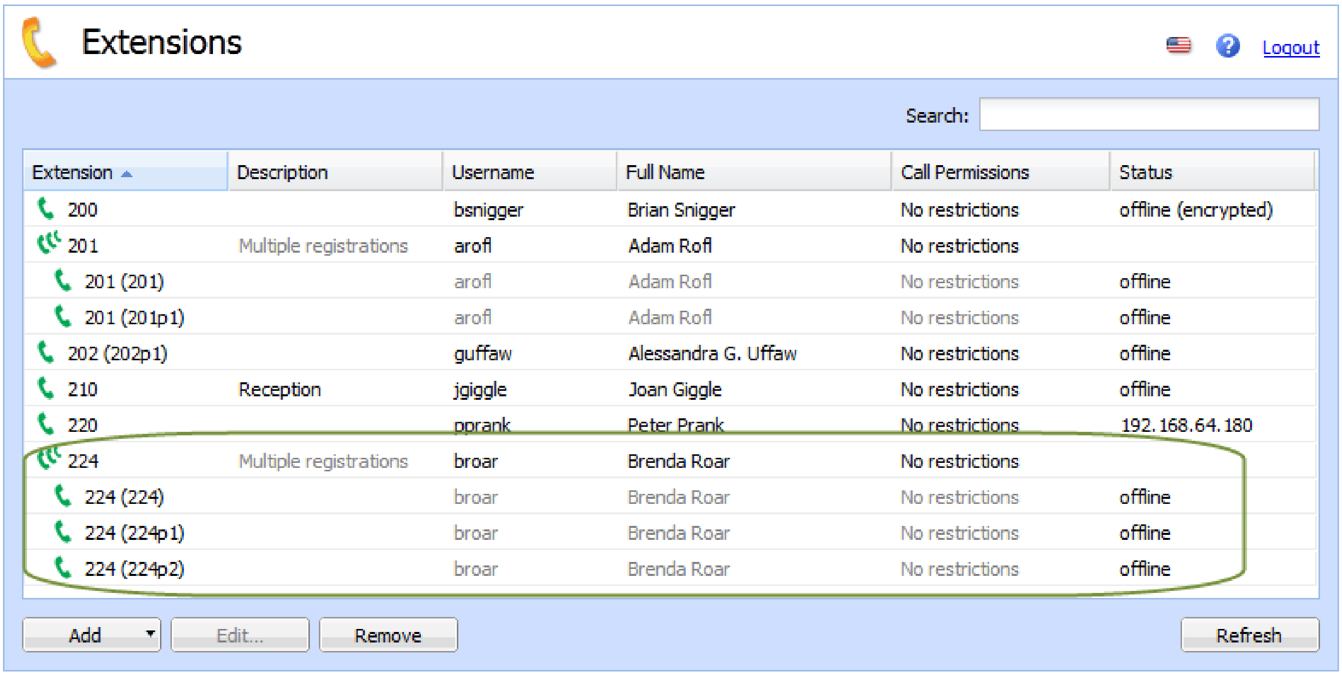
1. Open section **Configuration > Extensions**.

2. Select Brenda Roar's extension (224). Click on **Add > Add Another Registration**.



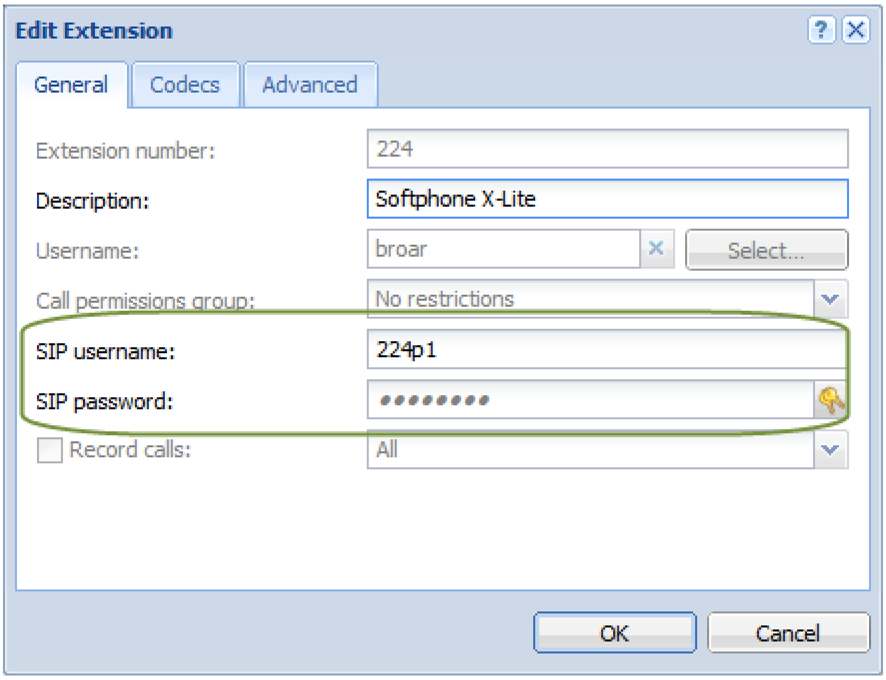
Screenshot 38: Extensions > Add Another Registration

3. A new registration is added to the user table. Add another registration. The result should be similar to the following image.



Screenshot 39: Extensions > Multiple registration

4. Double-click the `224p1` registration and note the SIP username and SIP password from the opened dialog.



Screenshot 40: Edit Extension > Login information for X-Lite

5. Click OK.

6. In the X-Lite settings (detailed information for installation can be found in topic Configuring the X-Lite software phone), enter the newly generated string into **User ID** and the SIP password into **Password**.

7. Repeat steps 4 to 6 for the second registration for the SIP client on a mobile phone.

4.3 Numbering

This section provides information about:

4.3.1 Mapping external and internal numbers

In Kerio Operator you can map external numbers to internal extensions. You can:

» Strip the first `0-n` digits from the number, including reducing the number to an empty string
» Add other digits to the beginning of the number

Routing incoming calls

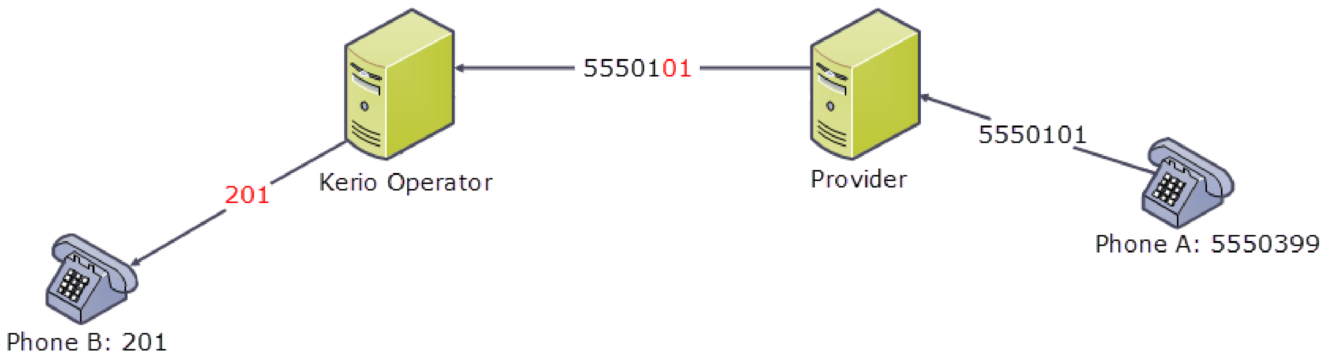
In Kerio Operator, you can use rewriting rules to map numbers for SIP and standard phone interfaces. Depending on your provider's requirements, you may need to strip out or change numbers

Example:

- A company has 100 phone numbers from a telephone provider.
- For incoming calls, the provider sends the whole number.
- For outgoing calls, the provider requires the whole number.
- Internal extensions have the format `2xx`.
- The prefix for outgoing calls is `9`.

When external **Phone A** (with the number **5550399**) calls internal **Phone B** (with the number **5550101** and the internal extension **201**):

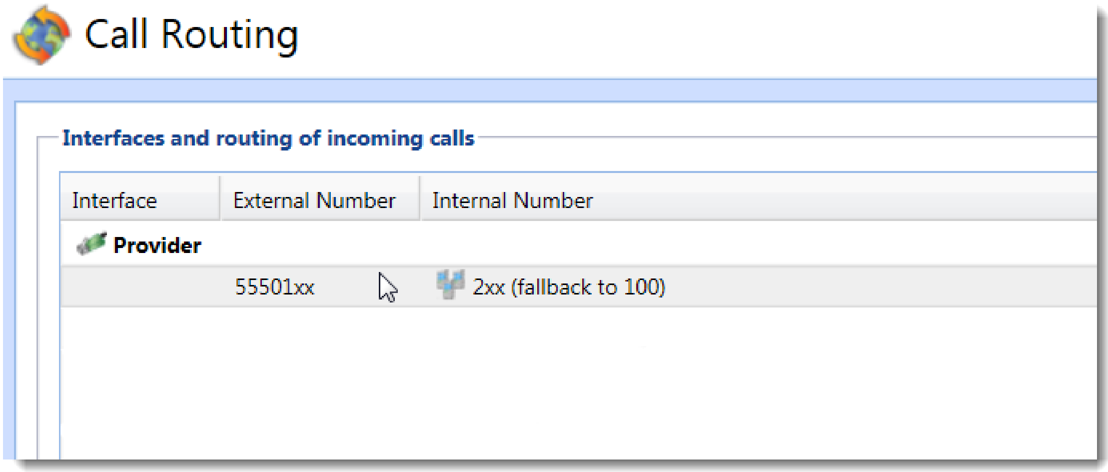
1. **Phone A** dials **Phone B's** number and a signal goes to the provider.
2. The provider sends the number to Kerio Operator.
3. The rewriting rule strips five digits from the left and adds the prefix `2`.
4. The call connects.



Mapping a trunk of numbers

To set the interface for an interval of numbers (`55501xx` in this example):

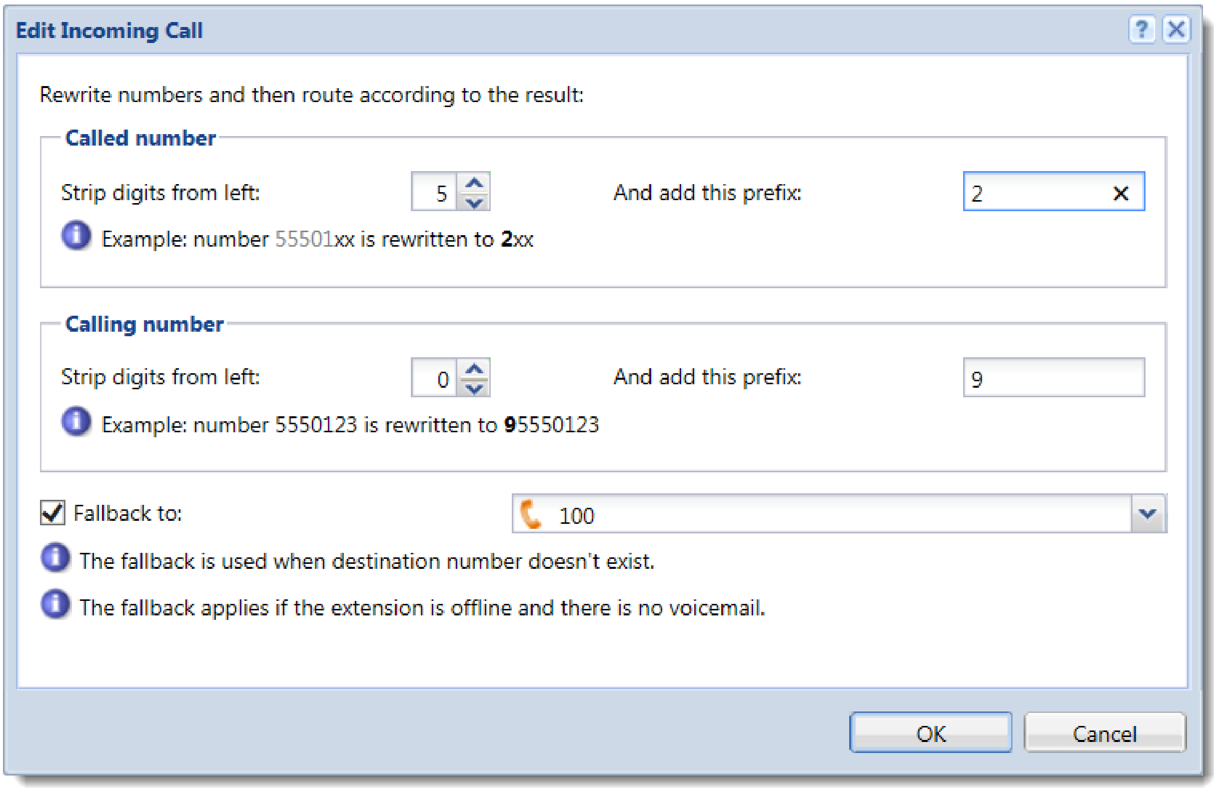
1. In the administration interface, go to **Configuration > Call Routing > Interfaces and routing of outgoing calls**.
2. Select the routing rule for the provider interface and click **Edit**. The **Edit Incoming Call** dialog box opens.



3. In the **Called number** section, strip the first five digits from the left, and add the prefix `2`. This modifies the number to the final format of the extension (`2xx`).

4. In the **Calling number** section, do not strip out any digits, and add the prefix `9`. This is useful when you want to call back the external number.

5. Click **OK**



Mapping a single number or multiple numbers

To set the interface for single or multiple numbers (`5550100 to 5550199` in this example):

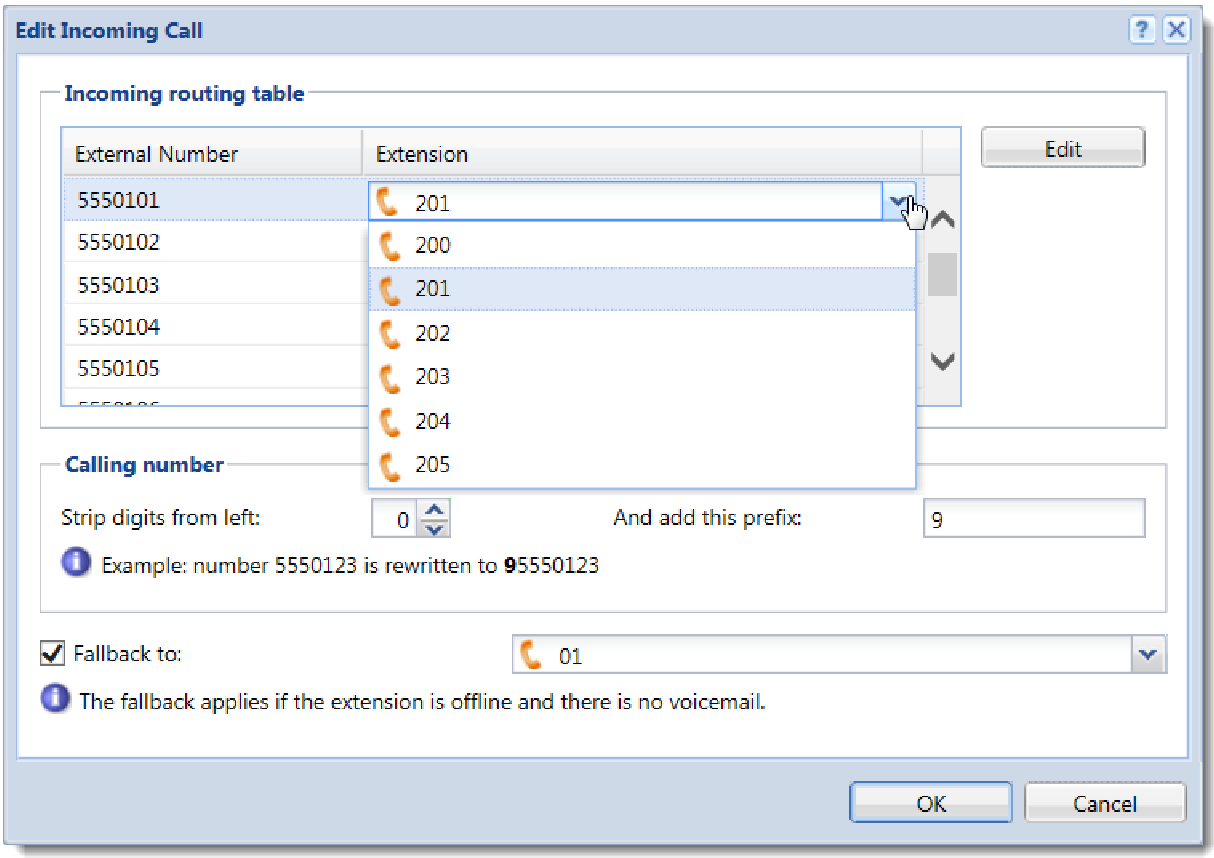
1. In the administration interface, go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.

2. Select the routing rule for the provider interface and click **Edit**. The **Edit Incoming Call** dialog box opens.

3. Double-click a line in the **Extension** column and assign an extension to the external number.

4. In the **Called number** section, strip the first two digits from the left, and add the prefix 2. This modifies the number to the final format of the extension (2xx).

5. Click **OK**



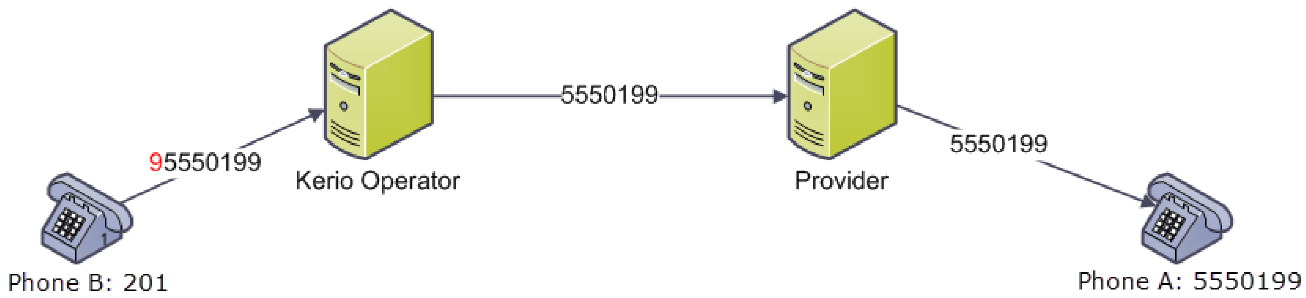
Routing outgoing calls

You can configure outgoing calls when creating an interface, either SIP or hardware cards.

For rewriting the numbers, you need additional configuration.

Example:

- External **Phone A** has the number **5550199**.
- Internal **Phone B** has the number **5550101** and the internal extension **201**.
- For outgoing calls, Kerio Operator uses the prefix **9**.
- The provider needs the whole number for outgoing calls.



When **Phone B** calls **Phone A**:

1. **Phone B** dials the number with the **9** prefix (**95550199**).

2. Kerio Operator uses rewriting rules and strips out the first digit (**9**). The number Kerio Operator sends to the provider is **5550199**.

3. The provider connects to **Phone A**.

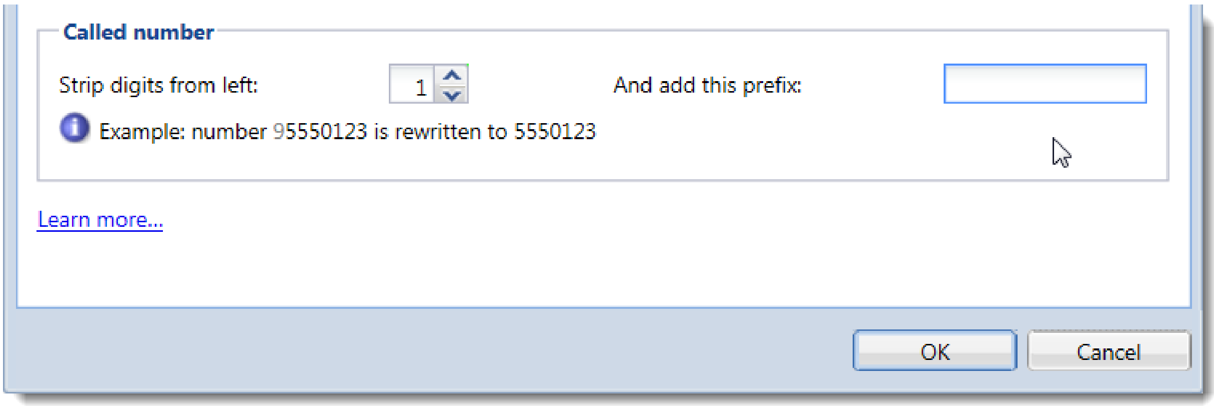
To achieve this configuration:

1. In the administration interface, go to **Configuration > Call Routing > Routing of outgoing calls**.

2. Select an interface and click **Edit**. The **Edit Outgoing Route** dialog box opens.

3. In the **Called number** section, strip one digit from left and do not add a prefix.

4. Click **OK**



Rules for outgoing calls

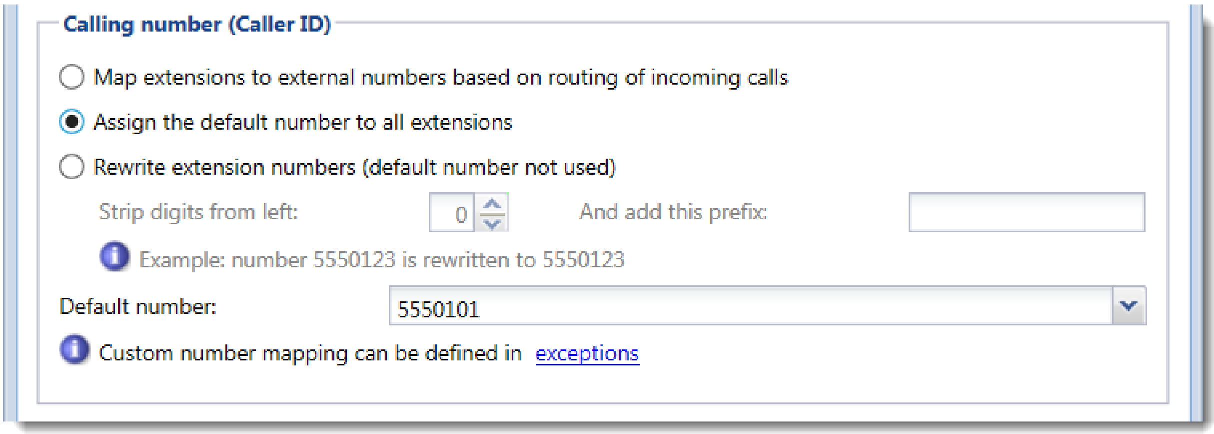
You can configure rules for outgoing calls:

1. In the administration interface, go to **Configuration > Call Routing > Routing of outgoing calls**.

2. Select an interface and click **Edit**.

3. In the **Calling number (Caller ID)** section, select one of these options:

- **Map extensions to external numbers based on routing of incoming calls** if you want to use a table of external numbers configured for the provider
- **Assign the default number to all extensions** if you want to use a default number for all extensions
- **Rewrite extension numbers (default number not used)** if you want to rewrite numbers in a specific way



Exceptions to the outgoing routes

To create an exception:

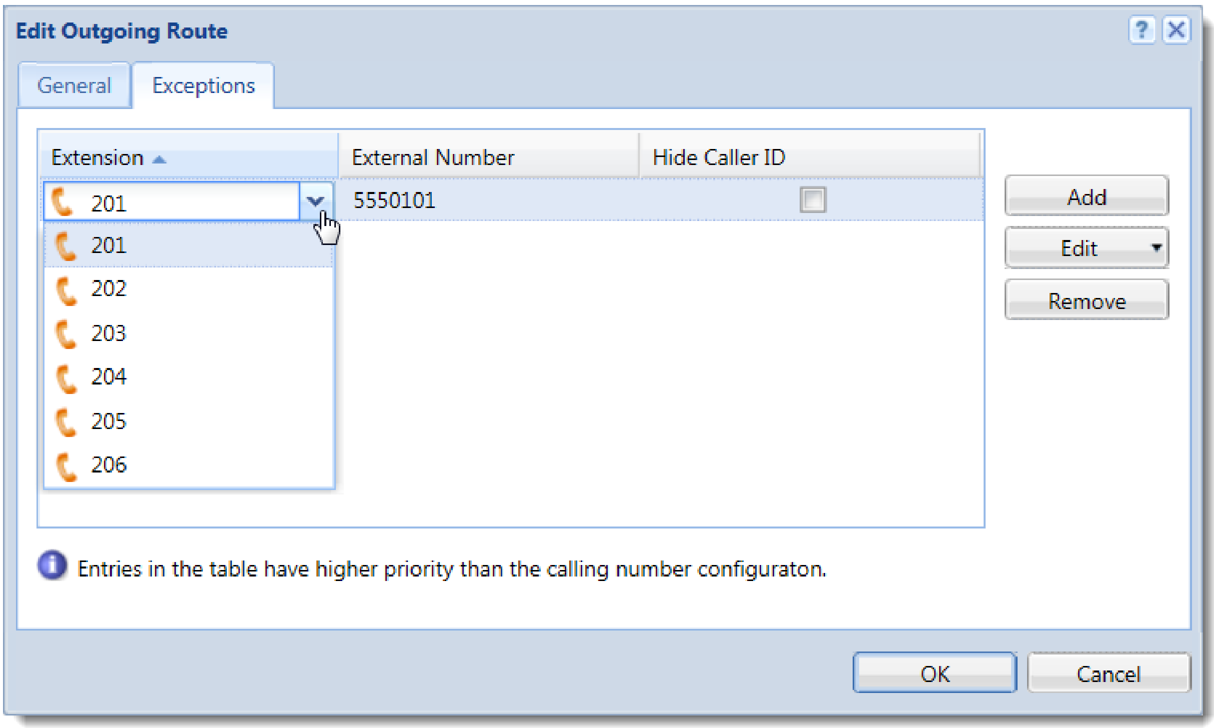
1. In the administration interface, go to **Configuration > Call Routing > Routing of outgoing calls**.

2. Select an interface and click **Edit**.

3. Enable the **Use route only for numbers defined in exceptions** option.

4. Click the **Exceptions** tab and click **Add**.

5. To change the internal number, double-click the displayed extension and select a new extension.



6. To change the external number, double-click the displayed number and select a new number.

7. If you want to hide this extension's number so the call recipient cannot see it, select the box in the **Hide Caller ID** column (see Displaying, hiding and overriding phone numbers for more details).

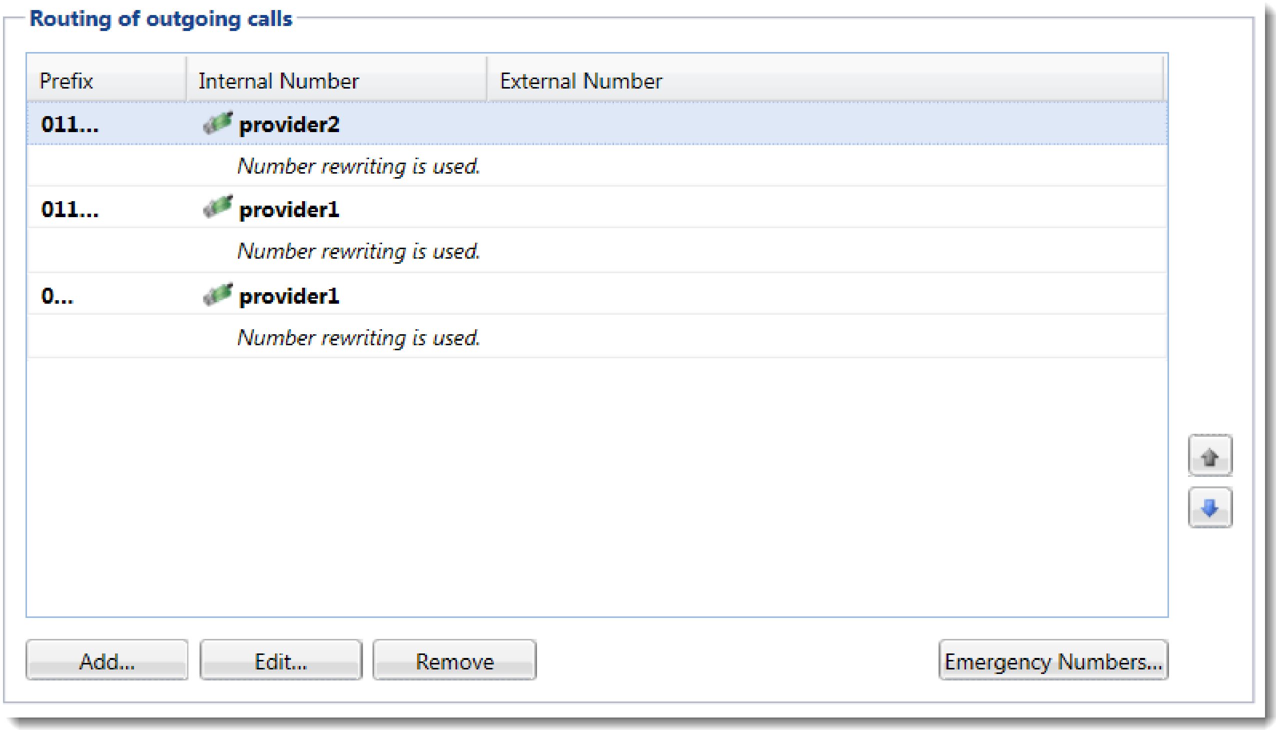
8. Click **OK**

Working with prefixes for outgoing calls

Kerio Operator works with prefixes for outgoing calls in a specific schema and you can use one prefix for multiple providers. Kerio Operator uses the longest prefix matching the dialed number. If that dial attempt fails, Kerio Operator tries the next route with the same prefix.

Example

- Use the prefix **011** for two providers (**provider1** and **provider2**) and the prefix **0** for outgoing calls.
- Dials the number **011 234 567**.

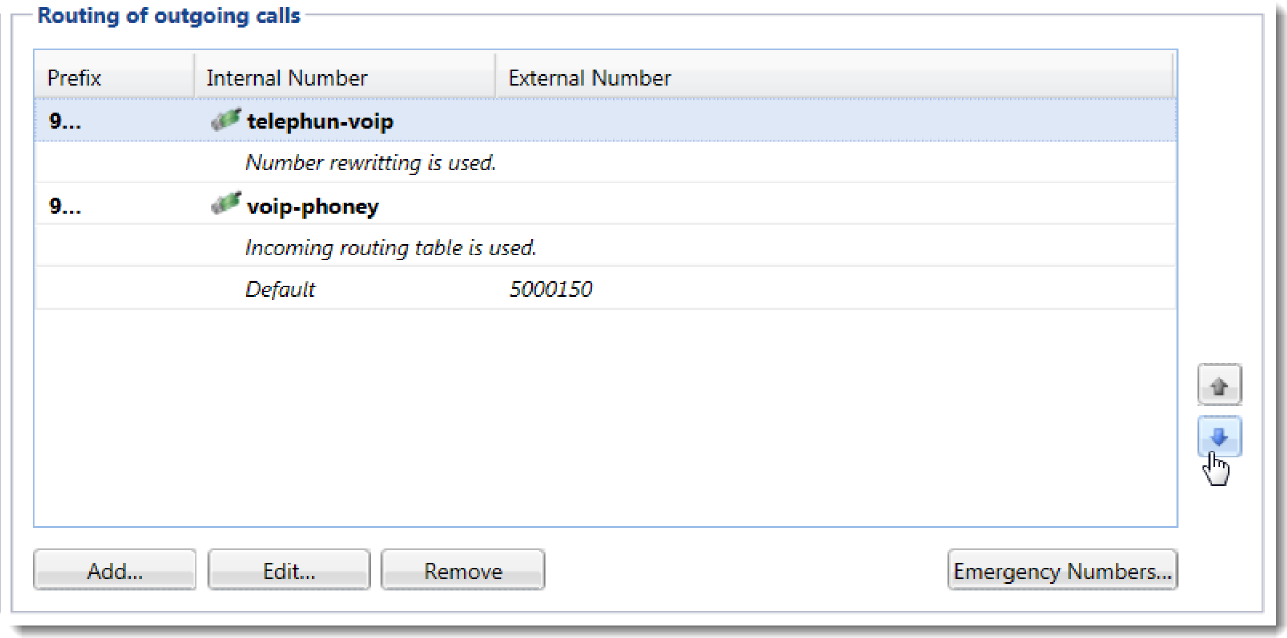


After dialing this number:

1. Kerio Operator goes through the **Routing of outgoing calls** table and tries to match the prefix.
2. Kerio Operator finds two matching prefixes, **0** and **011**, and uses the longest prefix.
3. Kerio Operator tries the **011** prefix to connect to **provider2**.
4. If the connection does not work, Kerio Operator uses the same prefix to connect to **provider1**.
5. If the connection still does not work, Kerio Operator does not try to use the last prefix (in this example, the **0** prefix), and the call fails.

Changing the order of prefixes

Kerio Operator works with providers for the same prefix in order from top to bottom. You can change that order by the using arrows on the right side of the administration interface to move it up or down.



4.3.2 Displaying, hiding and overriding phone numbers

Hiding users´ phone number

> **NOTE**
> Redesigned in Kerio Operator 2.4!

To hide users´ phone numbers for outgoing calls:

1. In the administration interface, go to the **Configuration > Call Routing > Routing of outgoing calls** section, select a prefix and click **Edit**. The **Edit Outgoing Route** dialog box opens.

2. Go to the **Exceptions** tab.

3. **Add** an extension.

4. Select the box in the **Hide Caller ID** column.

5. Click **OK**

> **NOTE**
> Some VoIP service providers do not allow hiding of phone numbers. If you use one of these providers, this settings do not work. See topic Connecting to VoIP service provider.

Changing phone number to a name

For outgoing calls, you can change the phone number to display a name:

1. In the administration interface, go to the **Configuration > Call Routing > Interfaces and routing of incoming calls** section, select an interface and click **Edit**. The **Edit External Interface** dialog box opens.

2. Go to the **Advanced** tab.

3. In the **Outgoing calls** section, select the **Override display name with** option, and type a new name.

4. Click **OK**

Extending display names for incoming calls

> **NOTE**
> New in Kerio Operator 2.4!

In Kerio Operator, you can extend the display name of incoming calls. The configuration works for all numbers that reach the interface and Kerio Operator adds the configured text to the beginning of the number or the caller's ID.

For example, a call center provides a technical support for several companies (for example, **Workplace**). Administrator wants to extend a display name of incoming calls with the company name, so the call center employees know from where comes the call:

1. In the administration interface, go to the **Configuration > Call Routing > Interfaces and routing of incoming calls** section.

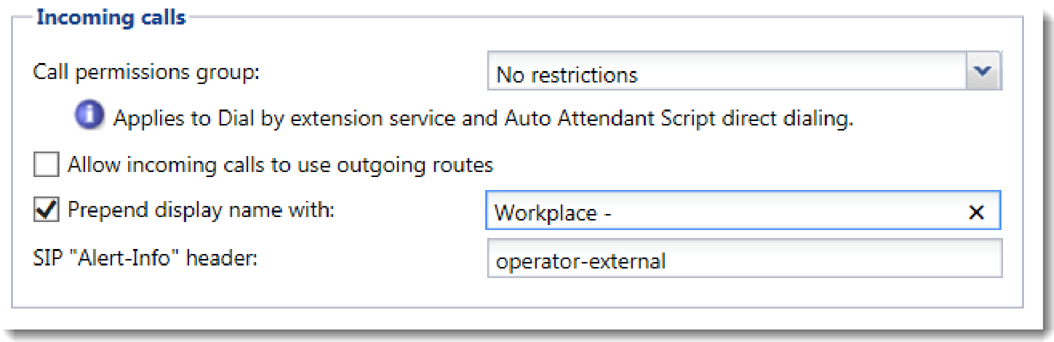
2. Select an interface and click **Edit**. The **Edit External Interface** dialog box opens.

3. Go to the **Advanced** tab.

4. In the **Incoming calls** section, select the **Prepend display name with** option, and type **Workplace -**.

5. Click **OK**

After this configuration, Kerio Operator extends all incoming calls to this interface with **Workplace -** (for example, **Workplace - 555 0155**).



4.3.3 Setting emergency numbers

When configuring emergency numbers, you can:

- add emergency numbers to the system,
- enable direct dialing (without the prefix for calling external networks).

> **NOTE**
> Call permitions and security restrictions are not applied to emergency numbers.

Configuring emergency numbers

1. In the administration interface, go to **Configuration > Call Routing**.

2. Click the **Emergency Numbers** button placed in the lower left corner.

3. Click **Overwrite** and select the country.

4. If the lists of emergency numbers do not suit your needs, click **Add** to create your own emergency numbers.

Enabling direct dialing

All outgoing calls to external networks use a prefix. You can configure an exception for emergency numbers:

1. In the administration interface, go to **Configuration > Call Routing**.

2. Click the **Emergency Numbers** button placed in the lower left corner.

3. Check **Enable direct dialing**.

4. Select **Used outgoing route**. This route will be used for all calls to the emergency numbers.

> **WARNING**
> If the direct dialing is enabled, you cannot create extensions which equal the emergency numbers.

4.3.4 Using number transformation

A number transformation in Kerio Operator ensures that phone numbers dialed automatically by an application (such as Click to Call) are dialed in the right format. The right format is the same format as for usual calls — without the outgoing prefix, or without the international call prefix. It depends on your SIP provider and their SIP server settings.

You may need the number transformation if your users use:

- Click to Call in Kerio Connect Client
- Kerio Operator App for Salesforce
- Kerio Phone

All the above mentioned applications dial phone numbers in the same format as they are displayed. If the number has an international prefix, Kerio Operator must delete it. If your Kerio Operator uses a prefix for outgoing calls, you must create a rule for adding the prefix in front of the phone number.

Configuring a number transformation

If you use an outgoing prefix in your environment, you must add a number transformation rule to Kerio Operator:

1. In the administration interface, go to **Integration**.

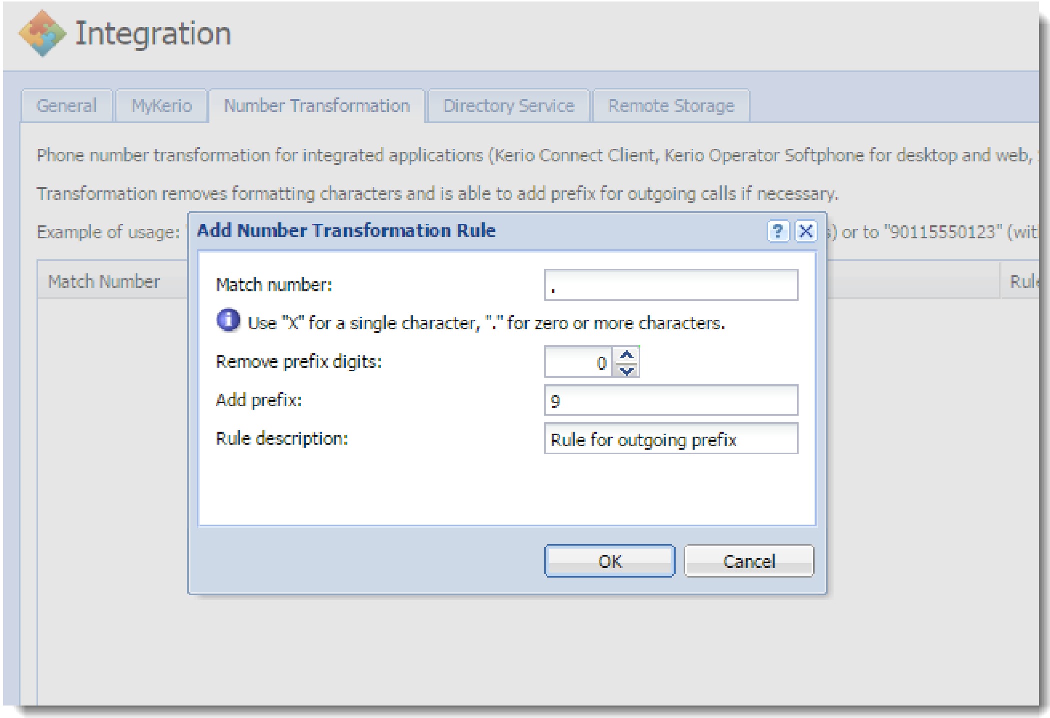
2. On the **Number Transformation** tab, add the rule for your outgoing prefix (for example 9).

3. Click **Add**.

4. In the **Add Number Transformation Rule** dialog, type dot in the **Match number** field. Numbers of any length are matched.

5. In the **Add prefix** field, add the outgoing prefix (for example 9).

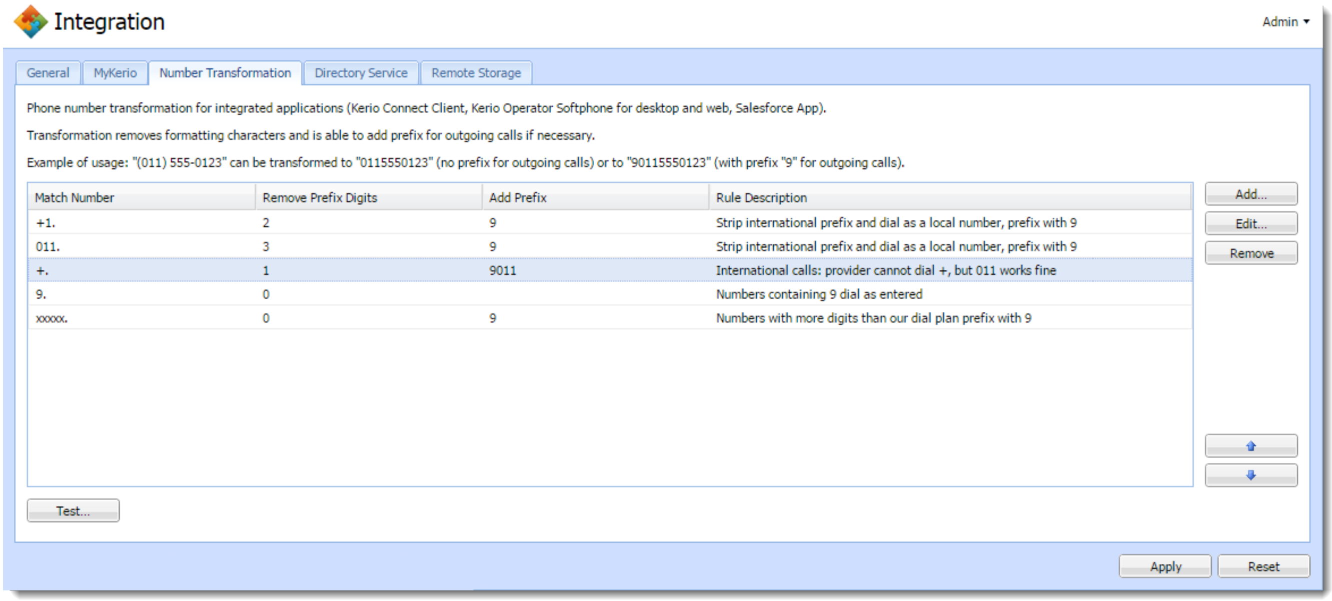
6. Click **OK**.



Example

The example uses the US international prefix and shows you a number transformation if:

- 9 is an outgoing prefix.
- Your SIP provider cannot dial numbers starting with a +.
- In case of local calls, you want to strip the international prefix.
- In case of international calls, you want to change + to 011.
- If the number does not start with 9, the rule adds 9 in front of the phone number.



4.3.5 Adding area codes to called numbers

In some situations you need to add an area code to your dialed numbers. In Kerio Operator, you can set the area code automatically.

Example:

- You use only 7-digit schema for your phone numbers (for example 555-5555)
- Your provider accepts only 10-digit numbers

To change your schema from 7-digit to 10-digit numbers:

1. Go to **Configuration > Call Routing**.
2. Click **Add...** under the **Routing of outgoing calls** section.
3. On the **General** tab, add prefix number (for example 9).
4. Select your interface.
5. For **Called Numbers** set **Strip digits from left** to 0 and type a 3-digit number prefix (for example 450).
6. Click **OK**

After that, all outgoing calls dialed with the 9 prefix have 10-digit format (in our example 450555-5555 instead of 555-5555).

Disabling outgoing calls to certain countries or regions

For more information, refer to Disabling outgoing calls to certain countries or regions (page 216).

4.4 Call settings

This section provides information about:

4.4.1 Bandwidth used by the different codecs

When you are using Voice over IP, the used VoIP phones and PBX can use different so called codecs. The consumption of bandwidth depends on which codec you use.

The following table gives you a short overview of the bandwidth consumption of the different codecs. On the page from asteriskguru.com, you can also find a bandwidth calculator tool.

codec	name	bandwidth (incl. overhead)	bandwidth for 5 concurrent calls	quality
G.711 a/u-law	PCM	80 kBit/s	512 kBit/s	ISDN
G.729	CS-CELP	32 kBit/s	200 kBit/s	good
iLBC	iLBC	32 kBit/s	200 kBit/s	good
G.723.1	MP-MLQ	21 kBit/s	110 kBit/s	average
G.723	A-CELP	15 kBit/s	80 kBit/s	average
GSM fullrate	GSM	13 kBit/s	80 kBit/s	average
G.726	AD-PCM	55 kBit/s	386 kBit/s	GSM
SpeeX	SpeeX	4 - 15 kBit/s	25 - 80 kBit/s	variable

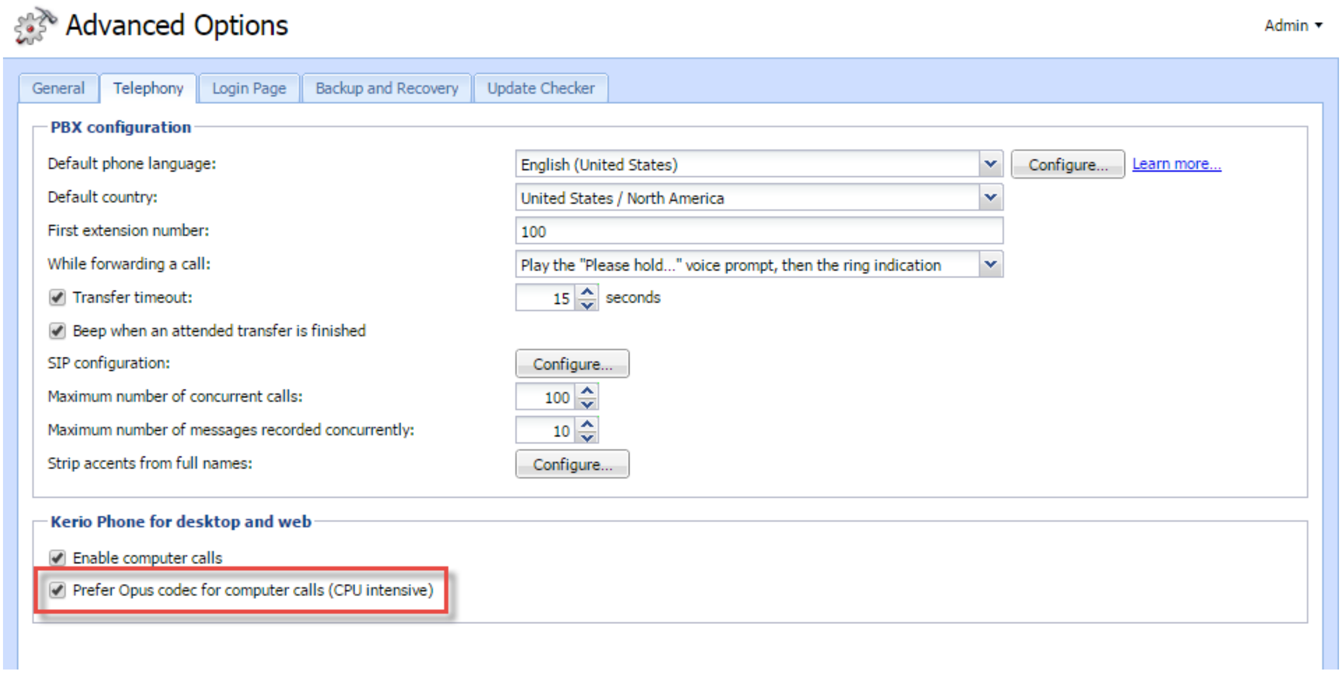
4.4.2 Using Opus codec for Kerio Phone

> **NOTE**
> New in Kerio Operator 2.5!

Kerio Operator allows you to use the Opus codec for calls via Kerio Phone for desktop and web. To use Opus for all your calls:

1. In the Kerio Operator administration interface, go to **Advanced Options** > **Telephony**.
2. In the **Codec configuration** section, select the **Prefer Opus codec** option.
3. Click **Apply**.

> **NOTE**
> Kerio Operator transcodes Opus to another codec every time the other caller doesn't use it. Transcoding calls increases the CPU usage. If you expect larger amount of concurrent calls, disable this option.



4.4.3 Redirecting calls

> **NOTE**
> Redesigned in Kerio Operator 2.4!

Kerio Operator can route incoming calls to different internal extensions or external numbers.

You can configure ringing rules (call forwarding) for each user in the **Ringing Rules** section.

> **NOTE**
> Users can also change their ringing rules in the Phone interface in the **Forwarding** section.

Configuring ringing rules in the administration

See the following example:

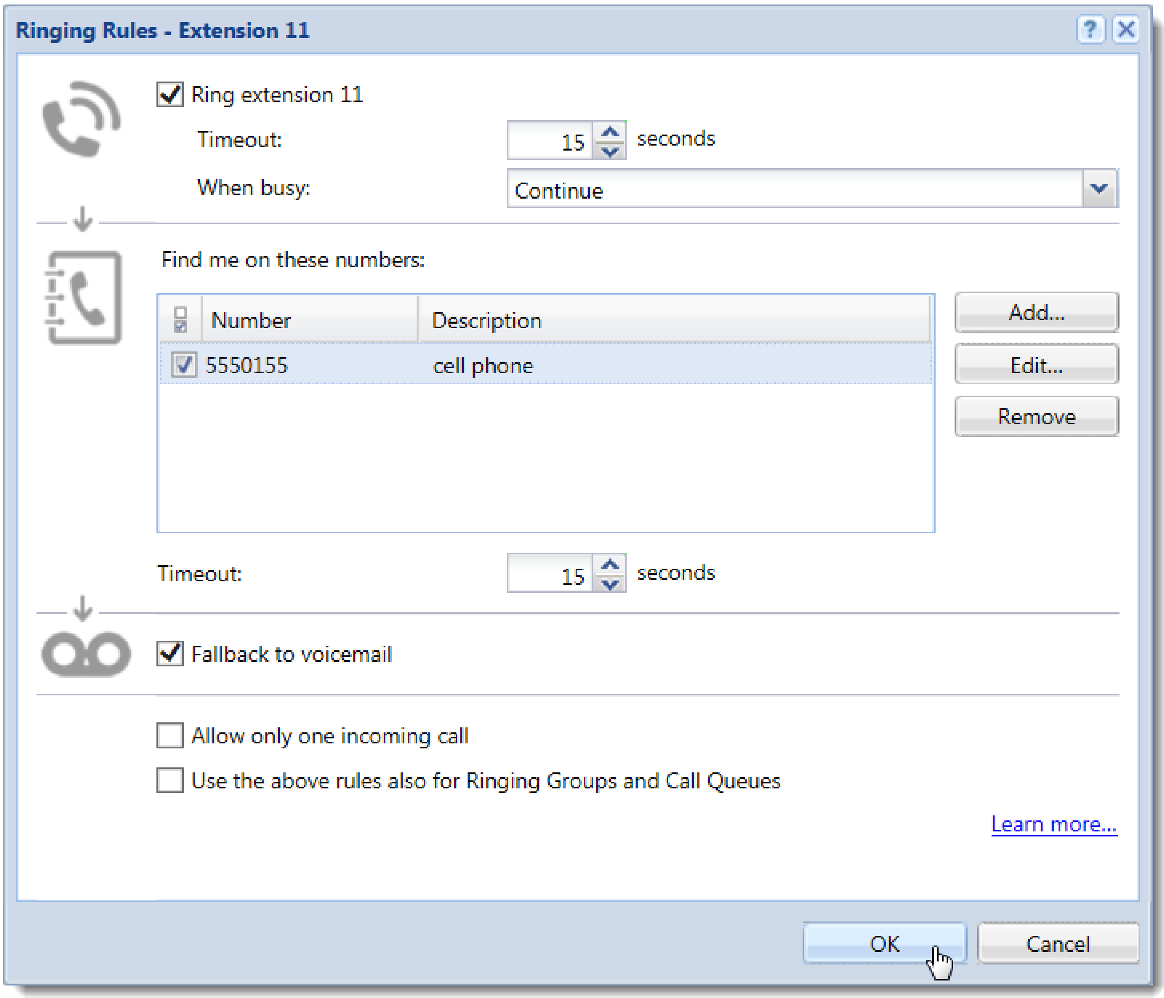
Bob has the internal extension `11` and a cell phone with the number`5550155`. He wants to receive calls on his cell phone. When he is busy, calls fallback to voicemail.

1. In the administration interface, go to **Configuration > Users**.
2. Select an account and click **Edit**. The **Edit User** dialog box opens.
3. Go to the **Extensions** tab.
4. Select an extension and click **Ringing Rules**.
5. Enable the **Ring extension** option.
6. Select a number for **Timeout**. When the specified time runs out, Kerio Operator forwards the call.
7. For **When busy**, select the **Continue** option.
8. Click Add and type the number **5550155** and a description (cell phone).

9. Select a number for **Timeout**.

10. Enable the **Fallback to voicemail** option.

11. Click **OK** to save your changes.



Additional configuration

> **NOTE**
> New in Kerio Operator 2.4!

For ringing rules, you can configure additional settings:

- Configure extension to allow only one incoming call
- Apply ringing rules to calls coming from call queues and ringing groups

Configuring extensions to allow only one incoming call

If your phones support multiple calls, you can configure your extensions to reject or redirect additional incoming calls when an extension is already busy with a call.

To allow only one incoming call at a time:

1. In the administration interface, go to **Configuration > Users**.

2. Select an account and click **Edit**. The **Edit User** dialog box opens.

3. Go to the **Extensions** tab.

4. Select an extension and click **Ringing Rules**.

5. Enable the **Allow only one incoming call** option.

6. Click **OK**

Kerio Operator now handles incoming calls using the configuration set in the **Ringing Rules** dialog box.

Applying ringing rules to calls coming from call queues and ringing groups

To configure ringing rules for calls from call queues and ringing groups:

1. In the administration interface, go to **Configuration > Users**.

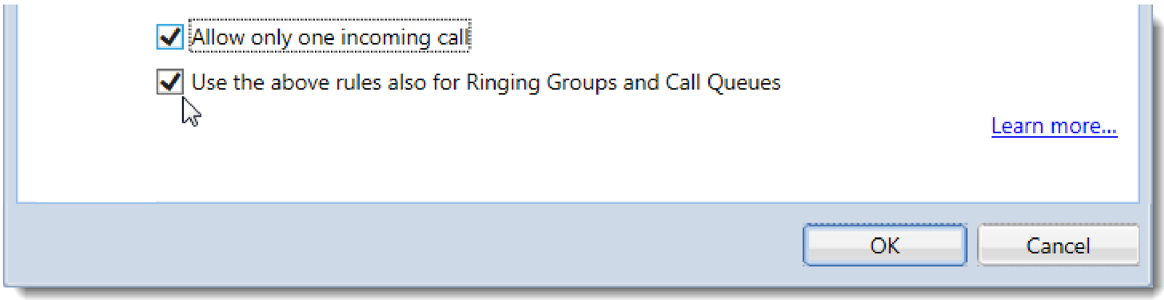
2. Select an account and click **Edit**. The **Edit User** dialog box opens.

3. Go to the **Extensions** tab.

4. Select an extension and click **Ringing Rules**.

5. Enable the **Use the above rules also for Ringing Groups and Call Queues** option.

6. Click **OK**



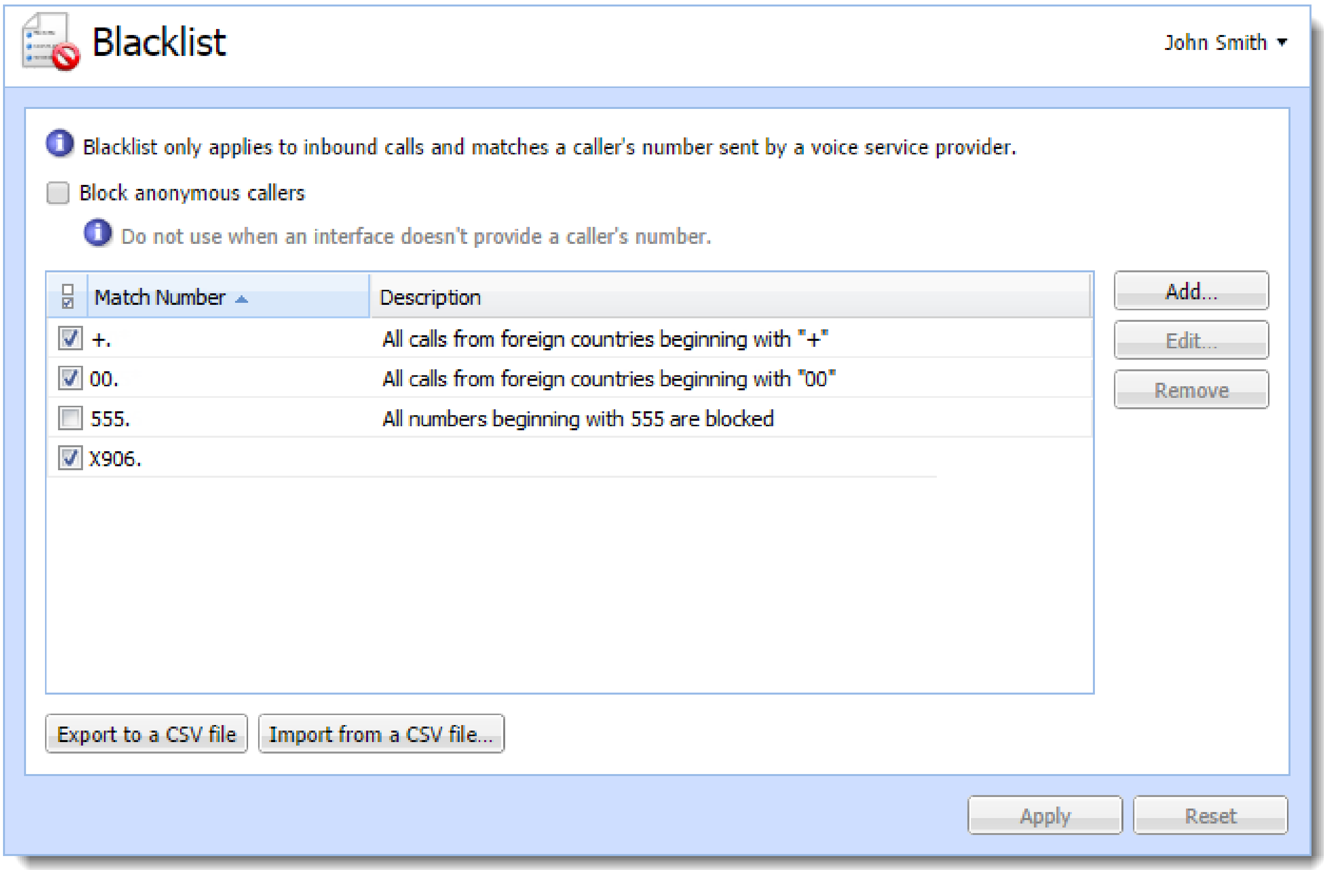
Configuring call forwarding in Kerio Phone

For more information refer to Redirecting calls in Kerio Phone.

4.4.4 Blocking incoming calls in Kerio Operator

If you want to block incoming calls from certain numbers, you can add the numbers to Kerio Operator's **Blacklist** .

Kerio Operator then blocks all numbers in the blacklist. No incoming calls from these numbers are connected.



Adding numbers to the blacklist

1. In the administration interface, go to **Configuration > Blacklist**.

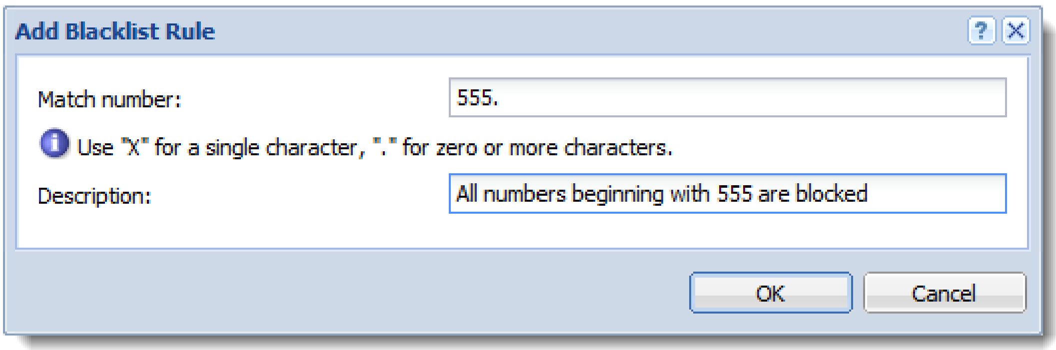
2. Click **Add**.

3. Type the number you want to block (**Match number**). You can match an entire number, or you can use X for single characters and . (dot) for multiple characters.

4. Add a description to document the reason for blacklisting the number.

5. Click **OK**

6. Add as many rules as you need.



7. (Optional) You can also block anonymous callers.

NOTE
Do not use this option if your provider does not show the caller's number. Otherwise, all incoming calls are blocked.

8. Click **Apply**.

When you receive a call from any of the numbers in the blacklist, your extension appears to be busy and the call is not connected.

Adding numbers from Call History

In **Call History**, you can select any incoming call number to add to the blacklist.

Right-click the number (a line) and select **Blacklist**.

When a call is blocked by blacklisting, you see **Blacklisted** in the **Status** column in **Call History**.

Adding/removing numbers with a PBX service

You can also use your phone to add numbers to the blacklist.

Kerio Operator has three pre-defined PBX services:

- `*30` for adding numbers to the blacklist
- `*31` for removing numbers from the blacklist
- `*32` for adding the last caller to the blacklist

To add a number to the blacklist:

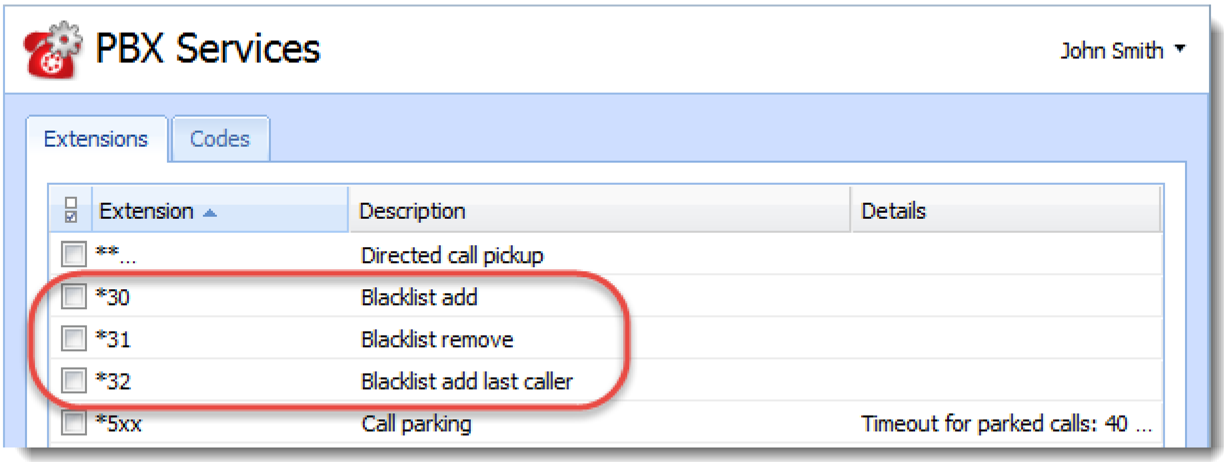
1. Dial the service number for adding numbers: `*30`.
2. After the beep, enter the phone number.
3. Hang up.

To add the last caller to the blacklist:

1. Dial the service number for adding last number: `*32`.
2. Confirm the number.
3. Hang up.

To remove a number form the blacklist:

1. Dial the service number for removing numbers: `*31`.
2. After the beep, enter the phone number.
3. Hang up.



Importing blacklists

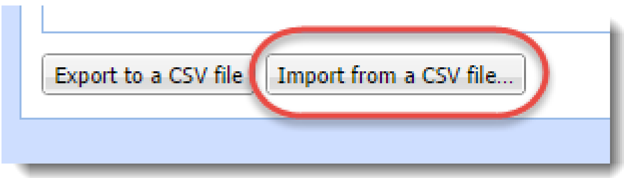
You can prepare a CSV file of numbers to be blocked and import init to Kerio Operator.

Each line in the file defines one entry. Entries must have the following format:

```
0,"555.","All numbers beginning with 555 are blocked"
1,"+.","All calls from foreign countries beginning with +"
1,"00.","All calls from foreign countries beginning with 00"
1,"X906.",""
```

> **NOTE**
> Separe all items with commas, and put number definitions and descriptions inside quotation marks.
> If any item is empty, keep the quotation marks.

To import the file, go to the **Blacklist** section and click **Import from a CSV file**.



Exporting blacklists

You can export the list of blacklisted numbers to a `*.csv` file.

1. Click **Export to a CSV file**.

2. Go to the correct folder, assign a file name, and save.

4.4.5 Disabling computer calls for Kerio Phone

> **NOTE**
> New in Kerio Operator 2.5.2!

Kerio Operator enables you to make calls via Kerio Phone for desktop and web using WebRTC.

You can disable the WebRTC support in the administration, so that users cannot see and use the **Computer** extension in their applications.

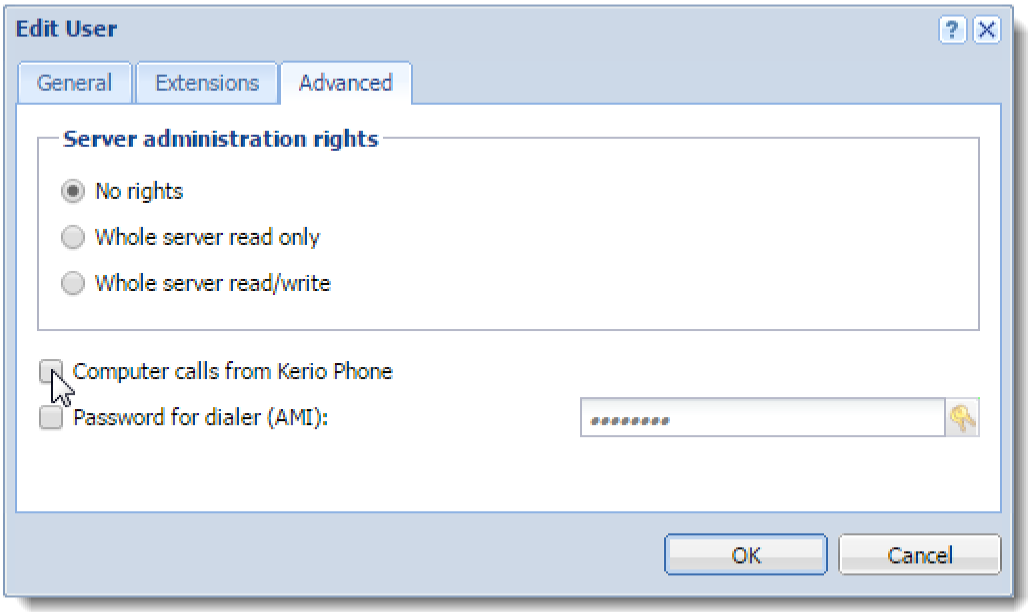
You can disable computer calls for:

- A single user
- Multiple users
- All users on your Kerio Operator Server

Disabling computer calls for a single user

1. In the administration interface, go to **Configuration > Users**.
2. Select a user and click **Edit**.
3. Switch to the **Advanced** tab.
4. Deselect the **Computer calls from Kerio Phone** option.
5. Click **OK**

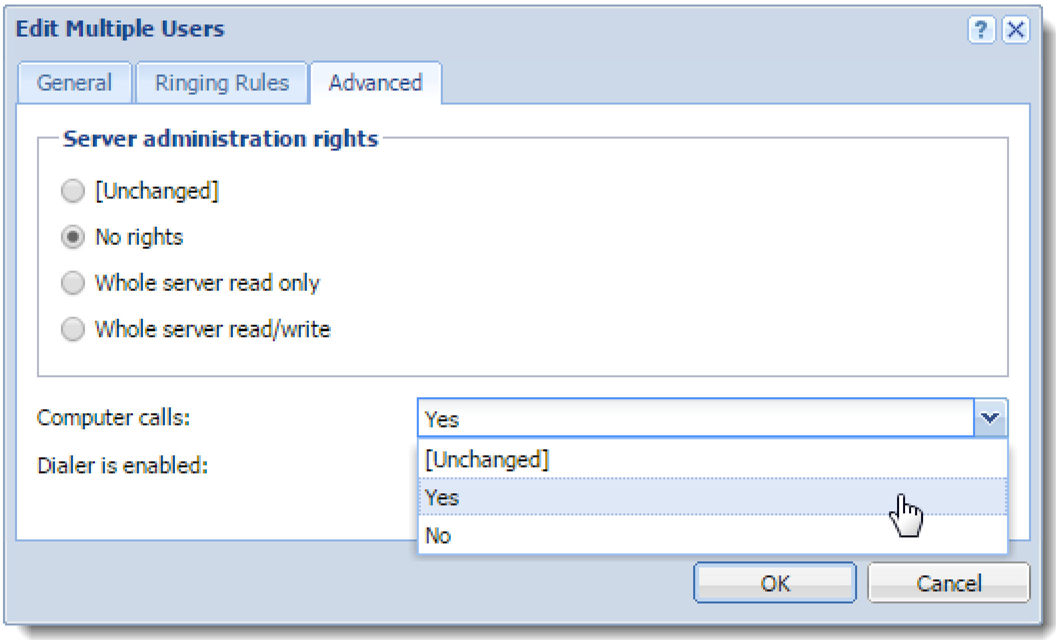
At this point, a selected user can no longer use the **Computer** extension for making calls.



Disabling computer calls for multiple users

1. In the administration interface, go to **Configuration > Users**.
2. Select multiple users and click Edit.
3. Switch to the **Advanced** tab.
4. In the **Computer calls** field, select **No**.
5. Click **OK**

At this point, selected users can no longer use the **Computer** extension for making calls.



Disabling calls for all users

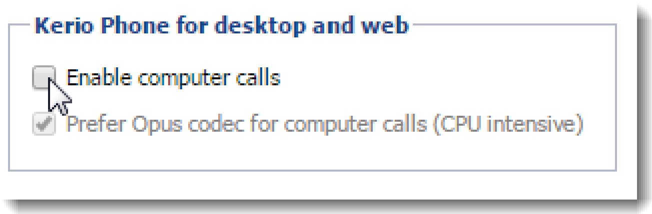
1. In the administration interface, go to **Configuration > Advanced Options**.

2. Switch to the **Telephony** tab.

3. In the **Kerio Phone for desktop and web** section, deselect the **Enable computer calls** option.

4. Click **Apply**.

At this point, users on your Kerio Operator server can no longer use the **Computer** extension for making calls.



4.4.6 Disabling outgoing calls to certain countries or regions

For security reasons, disable calls to countries users never call, create call permission groups and assign them to extensions.

Call permission groups can:

» Allow everything and disable certain prefixes, or

» Disable everything and allow certain prefixes

Disabling outgoing calls

1. In the administration interface, go to **Definitions > Call Permission Groups**.

2. Click **Add** or select an existing group and click **Duplicate**.

3. In the **Add Call Permission Group** dialog box, type the name and a description for the group and click **Add**.

4. Type a specific string of numbers, and choose the option to allow or deny access.

> **WARNING**
> To limit outgoing calls, include the prefix for outbound calls (usually 9).

5. (Optional) Repeat steps 2 and 3 to add additional numbers.

6. Click **OK** to save the settings.

> **NOTE**
> Kerio Operator applies the calls permissions in order, one by one.

Assigning call permission groups to extensions

1. In the administration interface, go to **Configuration > Extensions** and assign the created call permission groups to individual extensions.

2. Select an extension and click **Edit**. The **Edit Extension** dialog box opens.

3. Select a **Call permissions group**.

4. Click **OK**

> **NOTE**
> To assign a call permission group for multiple extensions, select multiple extensions and click **Edit**.

Adding area codes to called numbers

For more information, refer to Adding area codes to called numbers (page 207).

4.4.7 Video calling in Kerio Operator

> **NOTE**
> New in Kerio Operator 2.4!

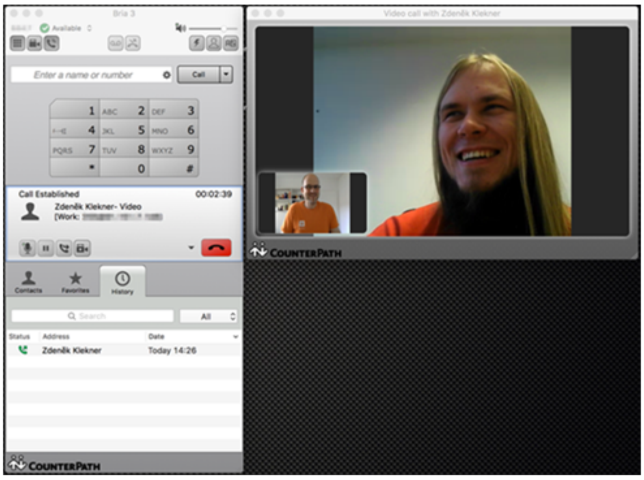
Kerio Operator now supports video calls with video enabled devices or software.

Prerequisites:

- Devices or software that use the same supported video codecs
- Configured extensions and interfaces to use the same video codec as your devices

Kerio Operator supports these video codecs (all are pass-through only):

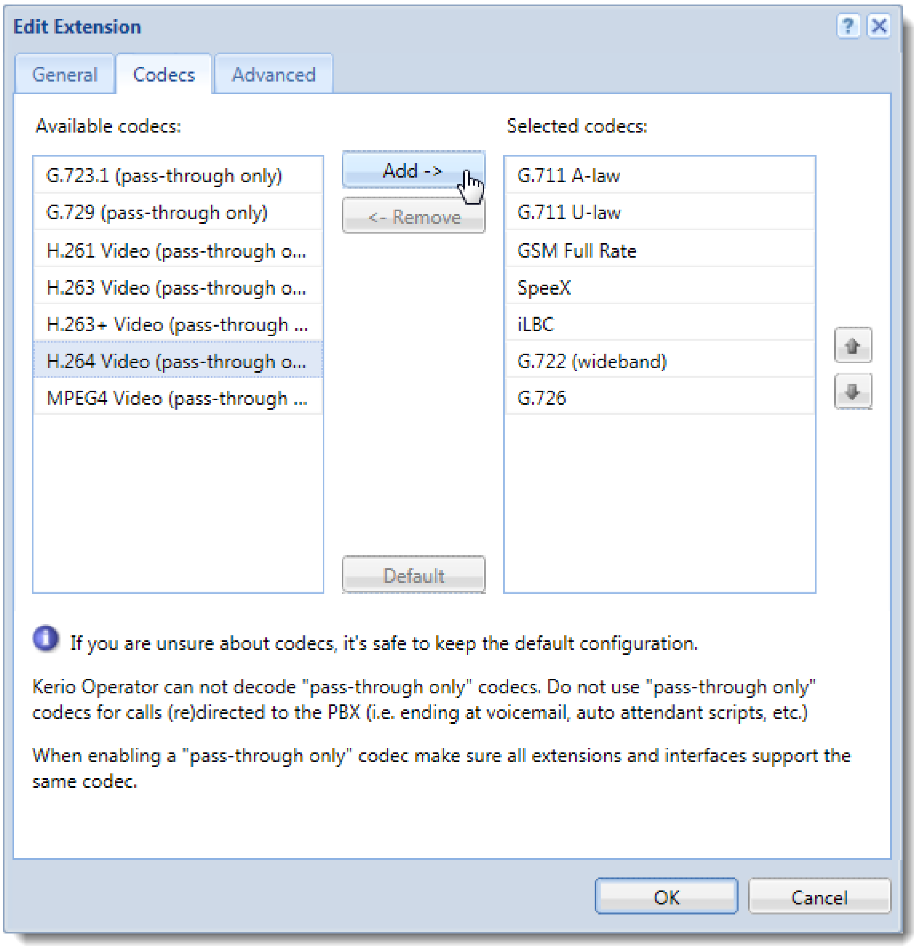
- H.261 Video
- H.263 Video
- H.263+ Video
- H.264
- MPEG4 Video



Adding video codecs to extensions

To enable video codecs for any extension:

1. In the administration interface, go to **Configuration > Extensions**.
2. Select an extension and click **Edit**. The **Edit Extension** dialog box opens.
3. Go to the **Codecs** tab.
4. Select a codec and click **Add** to insert the codec in the **Selected codecs** list.



5. Click **OK** to save your settings.

> **WARNING**
> All extensions participating in a video call must have the same codec.
> You can select a single codec and assign it to all your extensions.

Adding video codecs to interfaces

To enable video codecs for any interface:

1. In the administration interface, go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.
2. Select an interface and click **Edit**. The **Edit External Interface** dialog box opens.
3. Go to the **Codecs** tab.
4. Select a codec and click **Add** to insert the codec in the **Selected codecs** list.
5. Click **OK** to save your settings.

> **WARNING**
> Interfaces must have the same codecs as all extensions participating in a video call.

Troubleshooting

Video codecs in Kerio Operator are pass-through only and Kerio Operator cannot transcode them. For a proper connection, all devices must use the same codec. See the examples below:

Example of improper configuration

Device A tries to manage a video call with **Device B**:

- **Device A** works with the **H.261 Video** codec.
- **Device B** works with the **H.263+ Video** codec.

This configuration does not work, because the devices have different codecs and Kerio Operator cannot transcode them.

Example of proper configuration

Device A tries to manage a video call with **Device B**:

- **Device A** works with the **H.264 Video** codec.
- **Device B** also works with the **H.264 Video** codec.

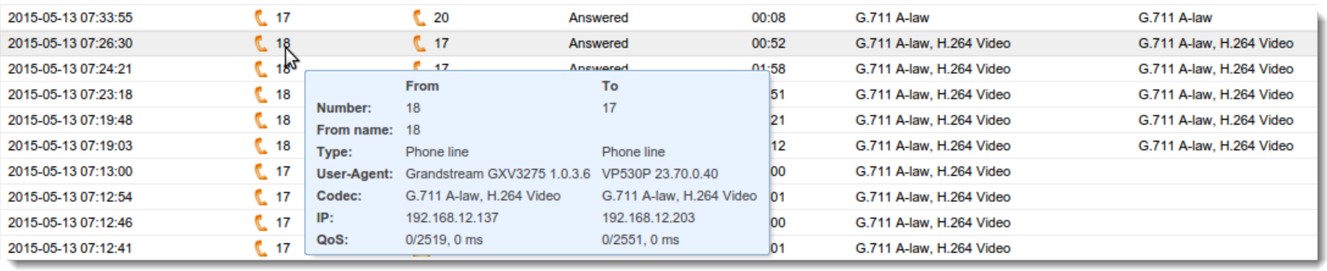
This configuration works, because both devices work with the same codec, so Kerio Operator does not need to transcode any codecs.

Phones do not display any video

If your phone does not display any video during the call:

- Set the same codecs for each device. To verify which codecs devices use, see the call history.
- Lower the resolution on the caller's phone.

For example, **Grandstream GXV3272** sends video call with 720p resolution to **Grandstream GXV3140**, but **Grandstream GXV3140** cannot decode the video. User decreases the resolution on **Grandstream GXV3272** and both phones start to display the video.



Phones do not transmit video

If your phone does not transmit video call, configure the device to make a video call.

For example, before you make the call, configure **Yealink VP-530** to prefer video calls.

Video is unstable

Devices with slow CPU or without a hardware acceleration can have problems with decoding the video:

» Decrease the resolution on the caller's phone.

» Verify that the network is not jammed. For example, transmitting a VGA signal using a **H.264 codec** takes `400 kbps` in each direction.

4.5 PBX services

This section contains information about configuring and using PBX services.

4.5.1 Using PBX services

Kerio Operator has special phone extensions which run the following services:

» Directed call pickup

» Call parking

» Call monitoring

» Call pickup

» Voicemail — a service extension to access voicemail. Kerio Operator recognizes which extension is used and you can set if PIN is required or not. This service is set automatically for provisioned phones.

» **Voicemail with login prompt** —a service extension to access voicemail. Kerio Operator is not able to recognize which extension is used. Users must authenticate with typing their extension and PIN.

» **Echo** — this option helps you monitor whether phones are correctly connected and what is the sound delay. Speak to the phone after hearing the automated message. If done correctly, your message is recorded and played back.

» **Music** — music plays upon dialing the extension (used for checking the connection).

» **Current time** — auto attendant tells the current date and time.

» **Dial by extension** — auto attendant invites the user to enter the extension which the operator will dial.

» **Dial by name** — user enters first several letters of the callee's surname and system searches among the users created in Kerio Operator and dials the extension.

» **Record audio** — Kerio Operator starts recording. Thus you can easily create records for auto attendant scripts in excellent quality.

» **Receive fax messages** — the service enables you to receive fax to defined email address. Necessary condition for enabeling the service is entering email address for receiving faxes in PDF format.

To configure PBX services, go to the administration interface > PBX Services.

> **WARNING**
> If you wish to use any service, tick the box next to this service. Extensions offering the services are disabled by default.

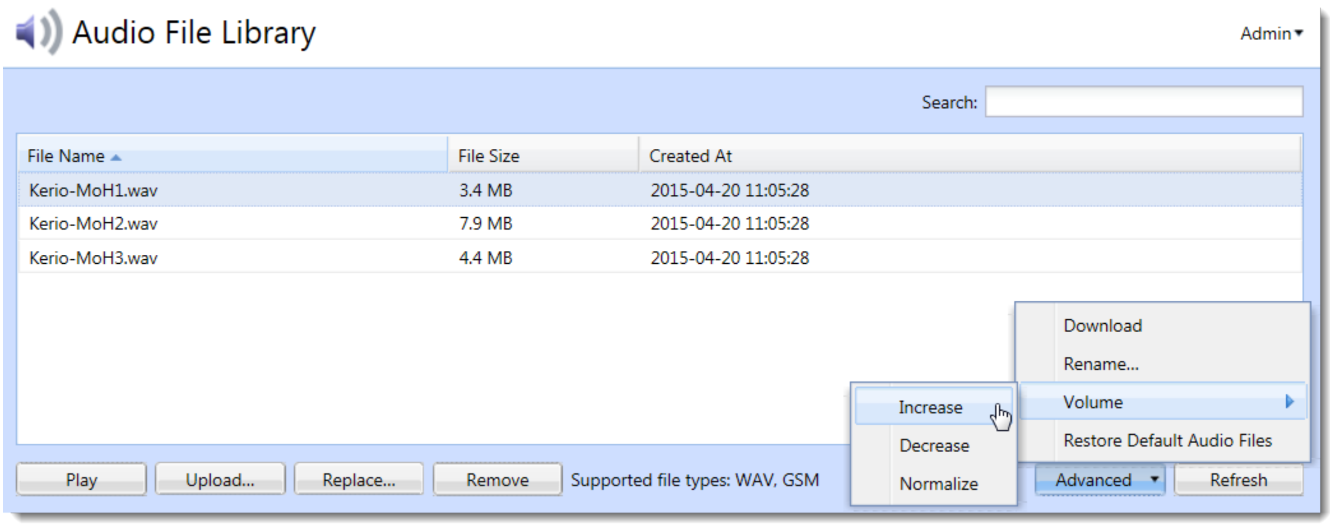
Creating voice files

This chapter shows how to create a records for an auto attendant script easily, fast and in sufficient quality.

1. Prepare texts.
2. In the administration interface, go to **PBX Services**, enable **Record audio** and save the settings.
3. Pick up the handset of your phone which is connected to Kerio Operator.
4. Dial the **Record audio** service.
5. Say individual voice recordings into the headset.

The record is stored in the audio file library in Kerio Operator. You can listen and manage the recordings in the **Definitions > Audio File Library**.

> **NOTE**
> If you open the administration interface in Safari browser and you cannot play any recordings, read topic Cannot play voicemails or audio files in Safari.

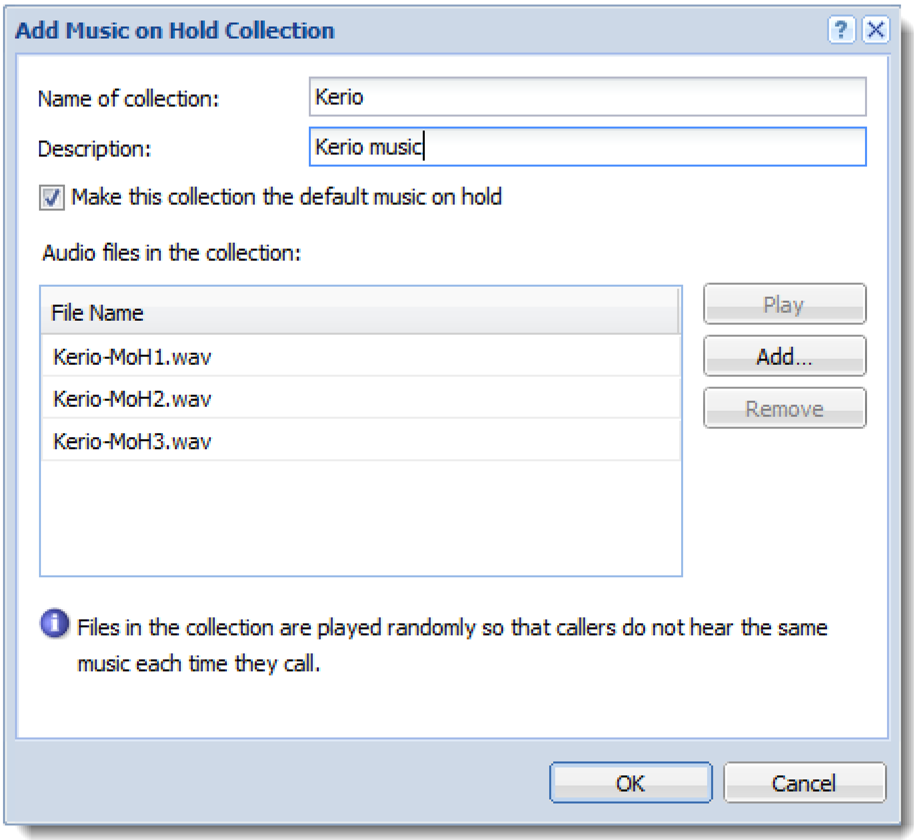


4.5.2 Configuring music on hold

While a caller is waiting for connection or in a call queue (see the Configuring call queues topic), they can hear recorded music. Kerio Operator has a default music collection. You can add and configure other audio files. You can upload any file in GSM and WAV format in section **Definitions > Music On Hold**.

Adding new collections

To add a new music collection (with one or more file), follow these instructions:



Screenshot 41: Adding New Collection

1. Go to **Definitions > Music On Hold** and click the **Add** button.

2. In the **Add Music on Hold Collection**, enter a name for the collection and a description.

3. Click the **Add** button situated on the right side of the table with added audio files.

4. In the **Select Audio File** dialog, add file one by one by clicking **Upload**.

5. Select a file in the list and double-click it. Repeat this step until all your uploaded files are listed in table **Audio files in the collection**.

Setting Default Collection

In the **Add Music on Hold Collection** dialog, check the **Make this collection the default music on hold** to ensure this collection is used as default in all other Kerio Operator Administration settings.

The default collection is used while holding the line (usually the **Hold** button on most phones). The other collections can be used, for example, in call queues.

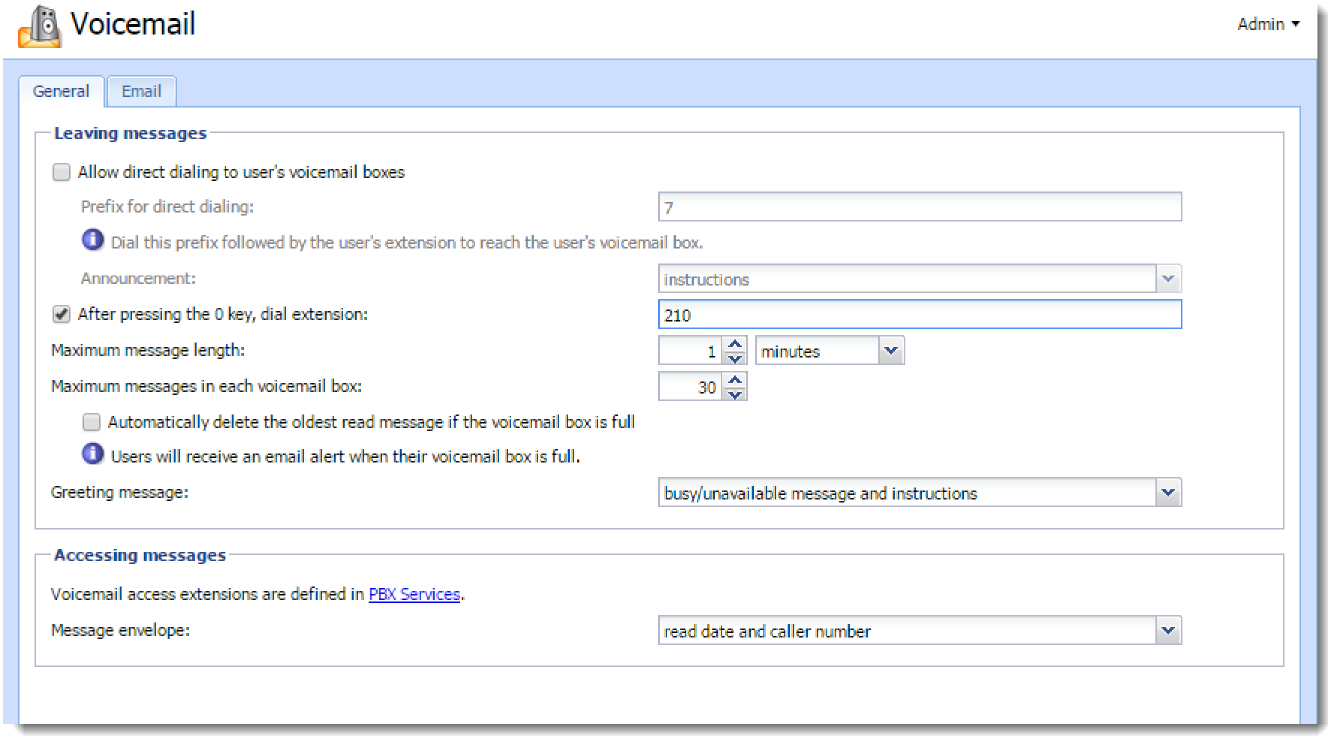
4.5.3 Configuring voicemail

Voicemail does not need any configuration. It works automatically once Kerio Operator starts. All users have forwarding to voicemail inbox enabled by default:

- when unavailable
- when busy

You can change the settings in section **Users** (Ringing rules). Users can also modify the settings in their Kerio Phone.

You can find the advanced voicemail configuration in the administration interface in section **Voicemail**.



Screenshot 42: Configuration > Voicemail

What is direct access to voicemail inbox and how to configure it

Direct access to users' voicemail enables the receptionist to connect calls directly to callee's voicemail.

1. In the administration interface, go to **Voicemail > General**.

2. Check **Allow direct dialing to user's voicemail boxes**.

3. Type a prefix in **Prefix for direct dialing**.

4. (Optional) Set an announcement (greeting message). If a call is redirected to voicemail, the caller hears a recorded message. This message can consist of two parts:

- **Instructions** inform callers what they should do next: Leave a message after the beep.
- **Message** informs callers that the callee is unavailable (the phone is switched off) or busy (the callee speaks with someone else).

5. (Optional) To change the size of users' voicemail boxes, adjust the value in **Maximum messages in each voicemail box**.

6. (Optional) To automatically delete read messages in full voicemail boxes, select **Automatically delete the oldest read message if the voicemail box is full**.

7. Click **Apply**.

Now the receptionist can dial the extension for direct access followed by the user's extension. The caller will be directed to the voicemail box of the person they are calling.

Enabling caller to escape voicemail by dialing 0

If you want to enable escaping voicemail by dialing 0, you must set an extension where the call is redirected:

1. In the administration interface, go to **Voicemail > General**.

2. Select **After pressing the 0 key, dial extension**.

3. Type an extension.

4. Click **Apply**.

Configuring forwarding of voicemail messages to user's email inbox

To send voicemail messages to email inboxes of the users, you need to set their email addresses in the administration interface in **Users**.

> **NOTE**
> If the users' INBOXes are unavailable (the mailserver is down), the user accounts are disconnected from voicemail and try to reconnect every 5 minutes. Each attempt to connect is recorded in logs.

My mailserver is Kerio Connect

For more information, refer to Integrating Kerio Connect and Kerio Operator (page 302).

My mailserver is a different SMTP server

1. On your mail server, create a special user which will be used for sending the voicemail messages. You can name them for example `operator`.

2. Go to administration interface to **Voicemail > tab Email** and check **Send each message to user's email**.

3. In **Mail server hostname**, type the SMTP server hostname and click **SMTP Configuration**.

4. Set the port number of the port used by your SMTP server. Usually 25 for SMTP and 465 for SMTPS

5. Decide, whether to communicate through secured connection. If the configuration of your mail server allows it, we recommend the encrypted connection to establish more secure communication.

6. If your SMTP server requires authentication, check **Server requires authentication**. Use the username and password for the account you created on your mail server in step 1.

7. Click **OK**

8. In **Voicemail > tab Email**, type a valid email address in **Sender email address** (so that your antispam rules accept it). The address should also represent the origin of the message. Example: `operator@live-and-let-laugh-inc.com`

Configuring the welcome message for callers

If a call is redirected to voicemail, the caller hears a recorded message. This message can consist of two parts:

Instructions inform callers what they should do next: Leave a message after the beep.

Message informs callers that the callee is unavailable.

How to set the greeting message?

1. Open section **Voicemail**.

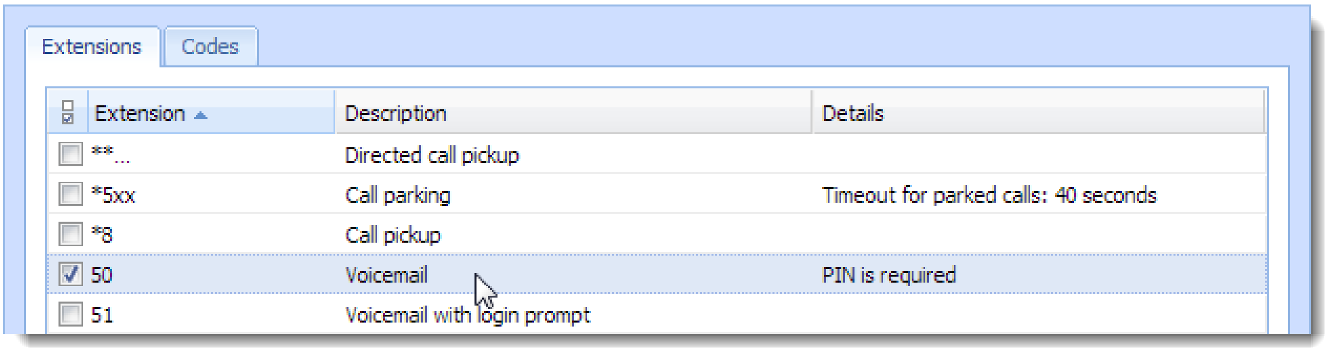
2. In the **Greeting message** menu, select whether the caller will hear the instruction, the message or both.

Greeting message for the direct dialing is described in the What is direct access to voicemail inbox and how to configure it section.

Changing the extension and voicemail PIN

Users use a special extension number to access their voicemail (by default: 50 or 51) and PIN.

To change the extension or enable/disable PIN, go to section **PBX Services** and read topic Using PBX services.



Screenshot 43: PBX Services

To set the user's PIN, go to account configuration in section **Users** to tab **Extensions**.

Accessing voicemail

- On your phone, press voicemail button or dial voicemail number and play the message.
- Through Kerio Phone.

> **WARNING**
> For users of Apple iPhone, iPad or Apple Mac OS X: If you cannot play your voicemail messages in Kerio Phone, contact the Kerio Operator administrator. An invalid certificate may be the reason.

- By forwarding voicemail to your mailbox (to get more information on this option, contact your network administrator).

Removing voicemail data for selected user

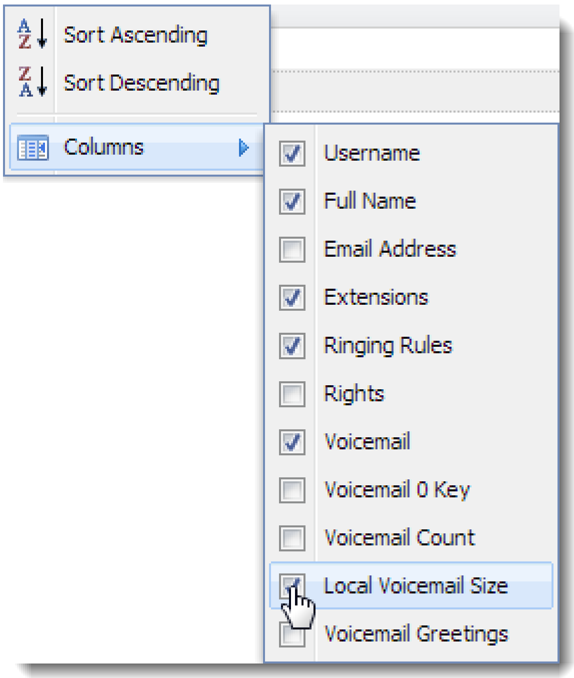
You can remove all local data connected with the particular user.

Local data is:

- voicemail
- custom voicemail greeting message

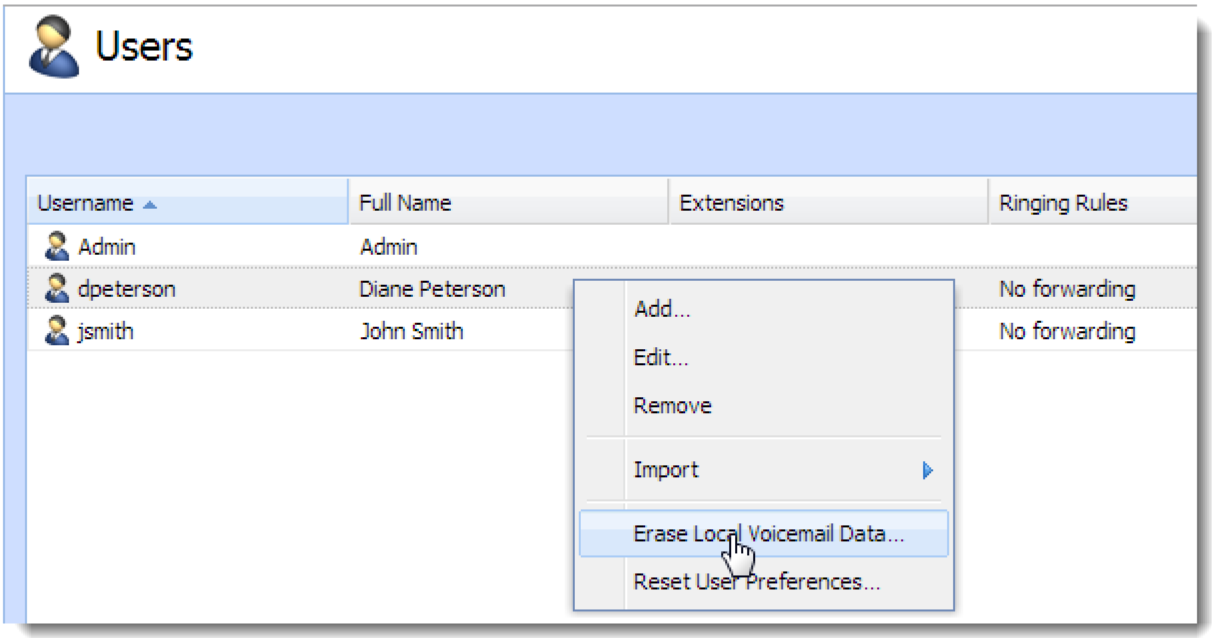
> **NOTE**
> Local data means that you cannot use this feature when you use the Kerio Connect integration — voice messages are stored in Kerio Connect in this case.

1. In the administration interface, go to **Users**.
2. Right-click the table heading.
3. In the context menu, select **Columns > Voicemail** and **Columns > Local Voicemail Size**.



Screenshot 44: Table context menu

4. Right-click the selected user and click **Erase Local Voicemail Data**.



Screenshot 45: User's context menu

If you succeed, there is value `0B` in the **Local Voicemail Size** column.

Managing voicemail via Kerio Phone

For more information refer to Using Kerio Phone

4.5.4 Configuring and using call parking

Call parking is a special type of call transfers. Parked calls wait for the callee on a special number.

Configuring call parking

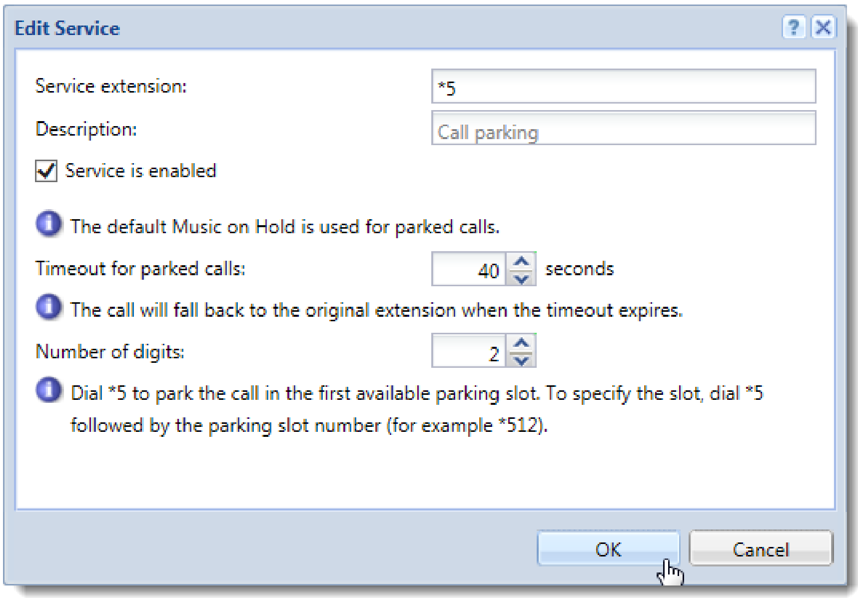
You can park calls on numbers which consist of:

- PBX service prefix
- Parking position number

1. In the administration interface, go to the **PBX Services** section.
2. Double-click **Call parking** to open the **Edit Service** dialog box.
3. Select the **Service is enabled** option.
4. In the **Service extension** field, type the call parking prefix. You can leave the default prefix setting *5.
5. Set the timeout (40 seconds by default). When the timeout expires, the call falls back to the original extension.
6. Set the number of digits for parking positions.

> **NOTE**
> se the same number of digits as for extensions (your dial plan). Users can park calls on positions which match their extension numbers.

7. Save your settings.



Using call parking

1. Initiate or answer the call.
2. Select the call transfer function on your phone. For more information, refer to Hardware telephone basic usage (page 126).
3. Dial the call parking number. You can:
   - Dial the whole parking slot number (for example, *512) to park the call to the specific slot.
   - Dial the **Call parking** extension only (for example, *5) to park the call in the first available parking slot. The voice-prompt message tells you the number of the first available parking slot.

4. Select the call transfer function on your phone.

5. Terminate the call.

To answer a parked call:

1. Pick up the phone.

2. Dial the call parking number (for example, `*512`).

If nobody answers the parked call before the timeout expires, the call falls back to the original extension.

4.5.5 Configuring and using conferences

Telephone conference is one telephone call of three or more users.

Telephone conferences allow participation of Kerio Operator users and external participants. To join a conference, participants must dial the conference number and PIN.

You can use two different types of conferences — statically or dynamically configured.

Statically configured conference

Statically configured means that conferences are created in the administration interface and each new conference uses one extension.

> **NOTE**
> If there is a lack of extensions, use dynamically configured conferences instead.

Configuring statically configured conferences

1. Go to section **Status > Dial Plan** and make sure that the extension you have selected for the conference is not used.

2. In **Configuration > Conferences**, click **Add**. The **Add conference** dialog is dispayed.

3. Enter the conference extension and its description.

4. In the menu **Conference type**, choose the **Statically configured** option.

5. Optional: Limit the number of participants.

6. Each conference can be protected by a PIN required from all participants upon attempting to enter the conference. If you wish to secure a conference, set a PIN and deliver it to the members.

7. To enable call recording, select **Record Calls**.

> **WARNING**
> Please note that call recording is a subject to special laws in many countries. It maybe illegal in your jurisdiction or require notice to the other party on the call. Accordingly, you assume all liability for using the call recording functions and are responsible for notifying all users of this system of this potential restriction, if applicable.

Connecting to a statically configured conference

1. Dial the conference telephone number / extension.

2. If the conference is protected, you will be asked to enter the PIN.

To leave the conference, simply terminate the call.

Dynamic conferences

A dynamic conference is created on one extension only. Users set the conference number and PIN after dialing the extension or the whole telephone number. On one extension, users can set unlimited number of conferences with different conference numbers.

The disadvantage of dynamic conference is that user has to enter three numbers when dialing the conference (the extension, the conference number and the PIN).

Configuring dynamic conferences

1. Go to **Status > Dial Plan** and make sure that the conference extension is not used by a user.
2. In **Configuration > Conferences**, click **Add**. The **Add conference** dialog is dispayed.
3. Enter the conference extension and its description.
4. In the **Conference type** menu, choose option **Dynamic, created on demand**.
5. To enable call recording, select **Record Calls**.

> **WARNING**
> Please note that call recording is a subject to special laws in many countries. It maybe illegal in your jurisdiction or require notice to the other party on the call. Accordingly, you assume all liability for using the call recording functions and are responsible for notifying all users of this system of this potential restriction, if applicable.

Connecting to a dynamic conference

To connect to an existing conference, enter the conference number and PIN (if required).

Creating a dynamic conference

1. Dial the conference telephone number / extension.
2. Enter any number for the conference.
3. Set PIN (if required).
4. Communicate these access numbers (extension, conference number and PIN) to other attendees.

To leave the conference, simply terminate the call.

Where to monitor conference activities

All current calls can be viewed under **Status > Conferences**.

4.5.6 Configuring auto attendant scripts

Auto attendant script is a simple collection of voice menus, submenus and announcements and actions defined for each of them according to the caller's behavior. It can:

- connect to an extension or voicemail,
- play an announcement,
- navigate through menus and submenus.
- send any faxes to a configured email.

Menus can be recorded in various formats. Kerio Operator supports the following formats:

Supported formats	Audio format
gsm	8KHz
wav	8KHz, 16 bits per sample, mono (Kerio Operator encodes all WAV files into this format automatically)

Screenshot 46: Kerio Operator — supported audio formats

How to add new auto attendant script

See the following description of an auto attendant script as an example. Create a script which:

» starts after dialing extension 200,

» contains a voice menu with the following text: LOL! You have just reached the Live And Let Laugh company's hotline (fiendish laugh).

- For Sales Department, press 1.
- For Quality Assurance Department, press 2.
- For Technical Support Department, press 3.
- If you wish to speak to the receptionist, press 4.

The Sales Department manages two flagship products of the company. Therefore, two submenus (**Joke Lite**, **Laugh Home 2012**) are created.

» For **Joke Lite**, press 1.

» For **Laugh Home 2012**, press 2.

» If you wish to talk to the receptionist, press 3.

Create the same menu for technical support.

Before creating the script, it is necessary to create extensions (in the assigned range 123456XXX) which will be used in the script.

» **extension 100** — reception of Live And Let Laugh Inc. One of the receptionists Joan Giggle or Brian Snigger will connect the calls if the caller makes no selection from the menu.

» **extension 203** — Quality Assurance Department extension (David Jester).

» **extension 301** — common extension (you can create a call queue or a ringing group) for Joke Lite experts, such as Frederic Jovial, George Funpoker, Anne Kdotte.

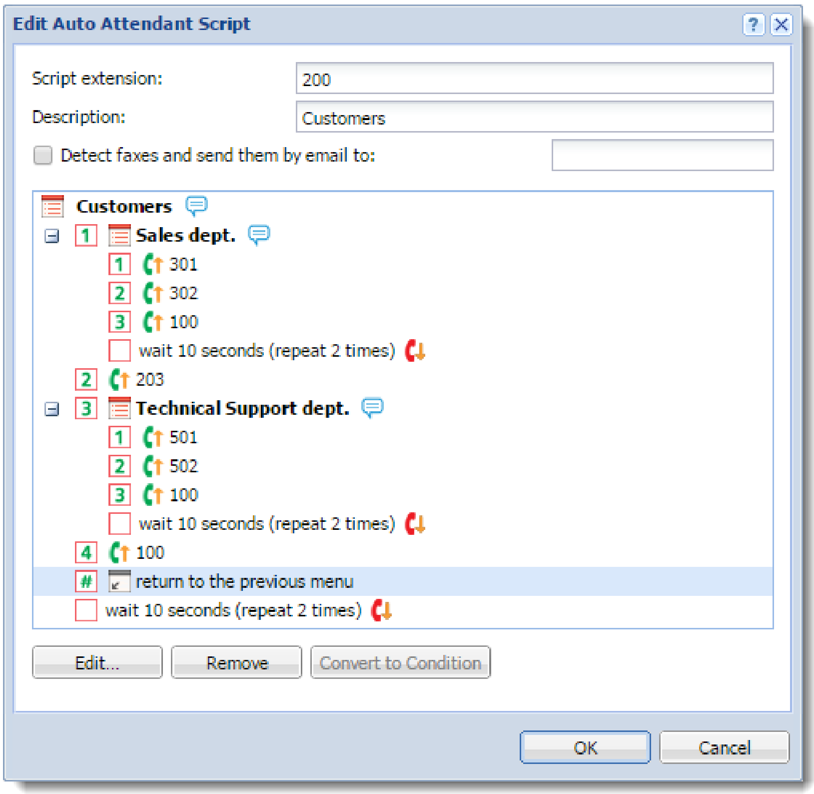
» **extension 302** — common extension for Laugh Home 2012 experts (Tamara Bellylaugh, Otto Spass, Mary Merry).

» **extension 501** — call queue for Joke Lite technical support (Andrew Widegrin).

» **extension 502** — call queue for Technical Support of Laugh Home 2012 (Alan Tickle).

Script settings

Configure the script in the administration interface in section **Configuration > Auto Attendant Scripts**:



Screenshot 47: Auto Attendant

1. Click **Add** and enter the **Script extension** (extension 200 in our example) and some description.

2. (Optional) To receive faxes to configured email address, select **Detect faxes and send them by email to** and type an email address.

3. Click **Edit** and open the **Edit Menu** dialog.

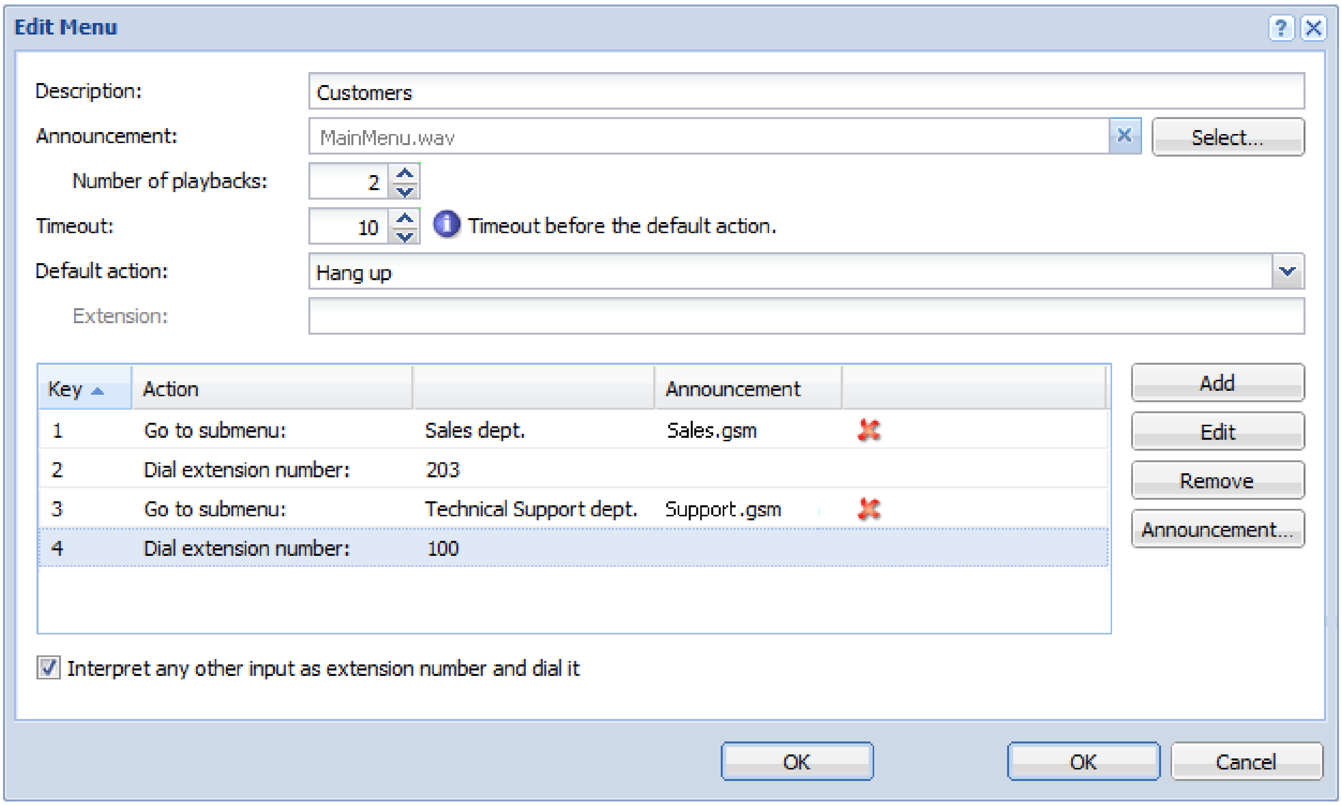
4. In the **Announcement** field, select the recording for the main script. The **Select** button offers existing recordings or you can upload your own announcement to the PBX.

> **NOTE**
> If you open the administration interface in Safari browser and you cannot play any recordings, read topic Cannot play voicemails or audio files in Safari.

5. Set **Number of playbacks** to two which will ensure the menu is played to the caller twice.

6. Once the announcement is played, timeout is started with the default action taken upon its expiration. Set the timeout to 10 seconds. The default action is the preset hang up action. This means that if the announcement is played twice and the customer does not make any selection within 10 seconds, the call will be terminated.

7. Click **Add**. The **Key** column states the key which confirms the customer's choice. Enter number 1. Enter 1 in this column. Column **Action** defines what happens when the caller presses a key on their phone. Select **Go to submenu**. We need to direct calling customers to the extension of the product they are interested in (either Joke Lite or Laugh Home 2012). In the **Announcement** column, you can add a record which will be played upon pressing the particular key (for example: Stay tuned, now you will be redirected to the Live And Let Laugh Inc Sales Department). Finish the table according to figure.

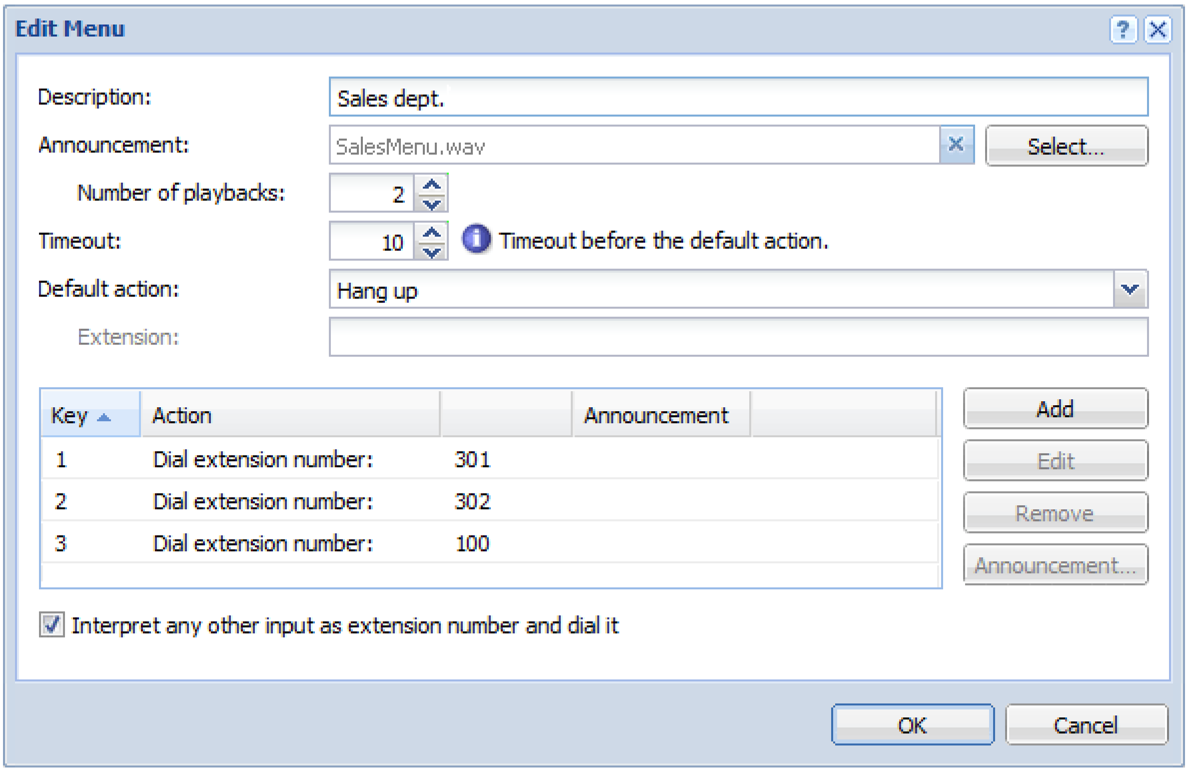


Screenshot 48: Editing main menu

8. Check **Interpret any other input as extension number and dial it**. This option allows to specify a direct extension while the auto attendant script is running.

9. Confirm the settings and return to the **Add Auto Attendant Script** dialog which is now similar to the one in picture above.

10. Click on menu **Sales dept.**. Again, the **Edit menu** dialog is opened but now the menu is for the Sales department. Follow the same procedure as with the main menu. The resultant menu will look as the one showed in the picture below.



Screenshot 49: Submenu edit

11. Do the same for the **Technical Support dept.** menu.

Now the script is complete.

> **NOTE**
> You can duplicate an existing script if you want to create a similar one — select a script and click **Duplicate**.

Time condition

The script can be limited to a specific time interval (office hours of your employees or night time when no call queue agents are available).

The time ranges (intervals) are configured in section **Configuration > Definitions > Time Ranges**. Once you have the time range configured, go back to the **Add Auto Attendant Script**, select the menu you wish to limit and click on the **Convert to Time Condition** button.

Instructions for time condition setting will be better understood through the following example focusing company's working hours. Sales department works from 9am to 5pm on weekdays. Configure the auto attendant script so that when customers call during office hours they will be connected to a sales department employee and when they call before or later they will hear a message announcing that the sales department is closed. To create the condition script, follow these instructions:

1. In the administration interface, go to **Configuration > Definitions > Time Ranges**.

2. Click **Add**. Dialog **Add Time Range** opens.

3. In section **Add to a group**, select the **Create new** option and enter a name for the new interval (for example, `Sales Department Office Hours`).

4. The **Description** is optional, for example **Weekdays from 9am to 5pm**.

5. Select `daily` in the **Type** menu and set the desired interval from 9 to 5 in the **From** and **To** fields.

6. In the **Valid on** menu, select **Weekdays**.

7. Click **OK** to confirm the changes.

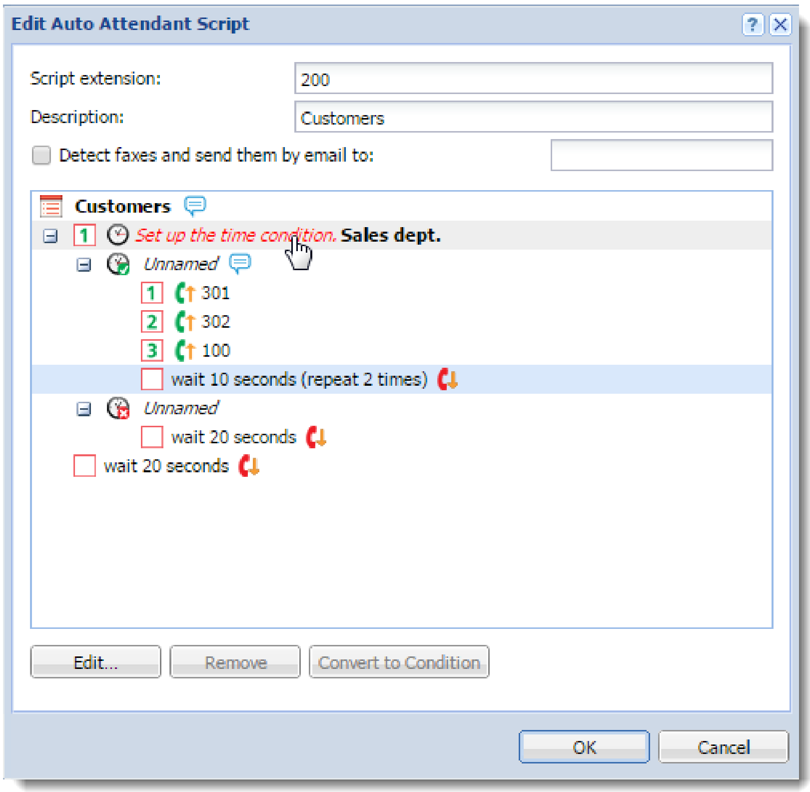
8. Open the **Configuration > Auto Attendant Scripts** section.

9. Click on **Add**.

10. In the **Add Auto Attendant Script** dialog, create a corresponding menu (the script created in the previous section will be used in this example — see the picture below).

11. Select the **Sales Department** submenu and click **Convert to Time Condition**.

12. Divide the Sales Department submenu in two time conditions. The first one is played if the condition is met and the second if the condition is not met. Click on the red highlighted text **Set up the time condition**.



Screenshot 50: Setting the time condition

13. This opens dialog **Edit Time Condition**. In the **For time range** menu, select **Sales Department Office Hours**.

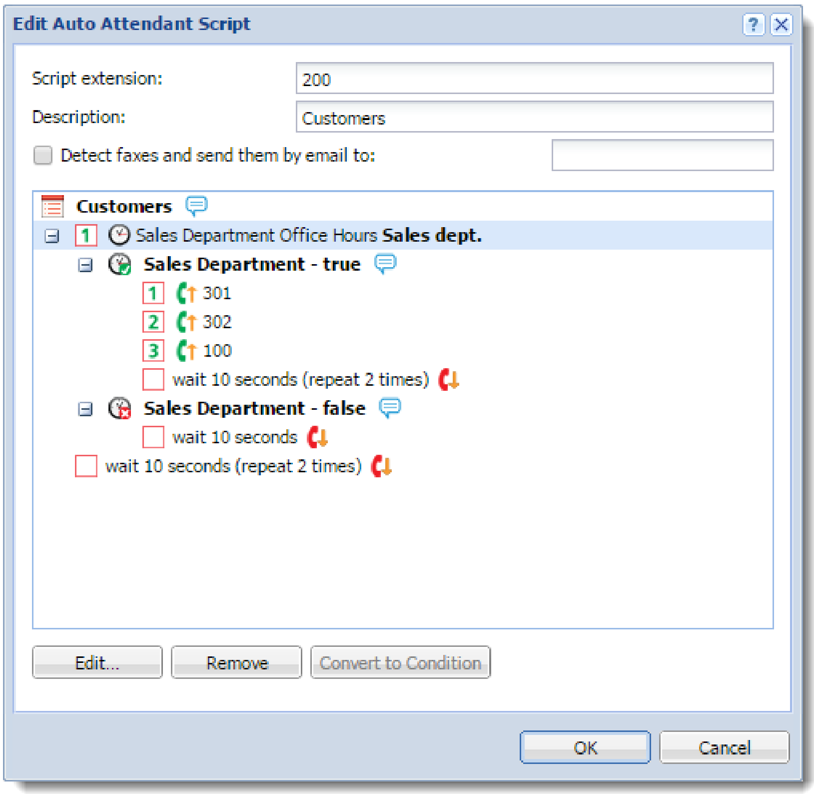
14. Click on the submenu representing the positive result of the condition. It is currently called **Unnamed**. In the dialog **Edit Menu** just opened, simply add a description (for example `Sales Department condition met`).

15. Click on the submenu representing the negative part of the condition (now it is empty and unnamed).

16. This opens dialog **Edit Time Condition** allowing to add a description (for example, Sales Department — condition not met).

17. Now you can modify the script. For example, in the **Announcement** field, add a message announcing that office hours of the Sales Department are from 9am to 5pm on weekdays.

18. Save the submenu. The resultant script is displayed in the next picture.



Screenshot 51: Time condition applied in the script

4.5.7 Setting time conditions in auto attendant scripts

Time conditions are best explained in an example

When configuring auto attendant scripts, Bob encountered the following problem. The company management created a new quality department. The responsible person is Alice. Bob created a new extension for this department. Alice came to Bob complaining that dissatisfied customers are calling constantly and she does not even have time for lunch.

Bob knew that Alice needs an auto attendant script which respects her working hours. And how to do it?

1. Bob created new time intervals for Alice's working hours, her lunch break and also for public holidays.

2. He created records for the following announcements:

- Hello. You are calling Live And Let Laugh Inc. We are having a delicious lunch at the moment. If you call after 1pm, we will gladly hear what you have to say. Talk to ya later!"
- "Hello. You are calling Live And Let Laugh Inc. We are off the clock at the moment. Please, call us on weekdays from 8am to 12pm or after lunch from 1pm to 6pm. We will gladly hear what you have to say. Talk to ya later!"
- "Hello. You are calling Live And Let Laugh Inc. Have a very merry holiday today. If you wish to make a complaint, call us on weekdays from 8am to 12pm or after lunch from 1pm to 6pm. We will gladly hear what you have to say. Talk to ya later!"

3. He created a new auto attendant script with time conditions.

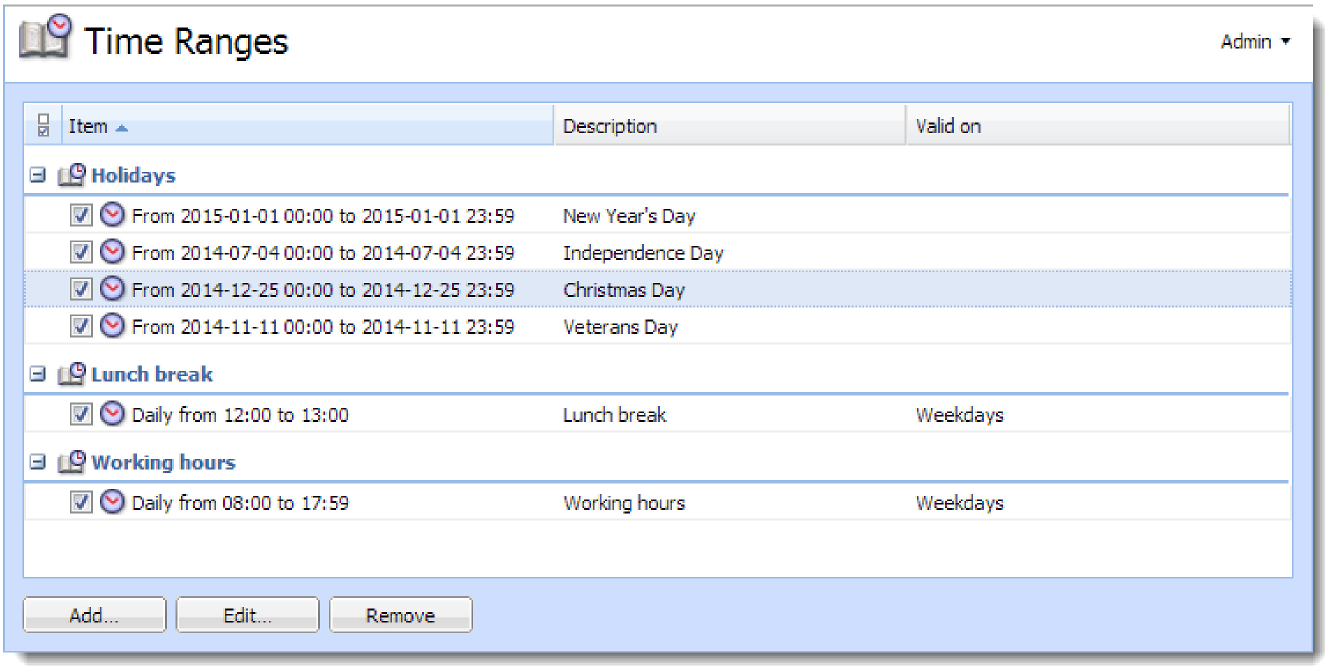
> **NOTE**
> You can also use the Day/night mode to create time conditions without a specific time set. For more information, refer to Using the Day/night mode in auto attendant scripts (page 241).

Setting time intervals for auto attendant scripts

1. In the administration interface, go to section **Definitions > Time ranges**.

2. Add three new time ranges. Two ranges will be of the daily time — `Lunch break` and `Working Hours`. Both ranges will be valid on weekdays.

3. The third range will be absolute. Add the first public holiday when creating the range. Add also other public holidays and do not forget to add them into the existing group `Holidays`.



Creating auto attendant scripts in Kerio Operator

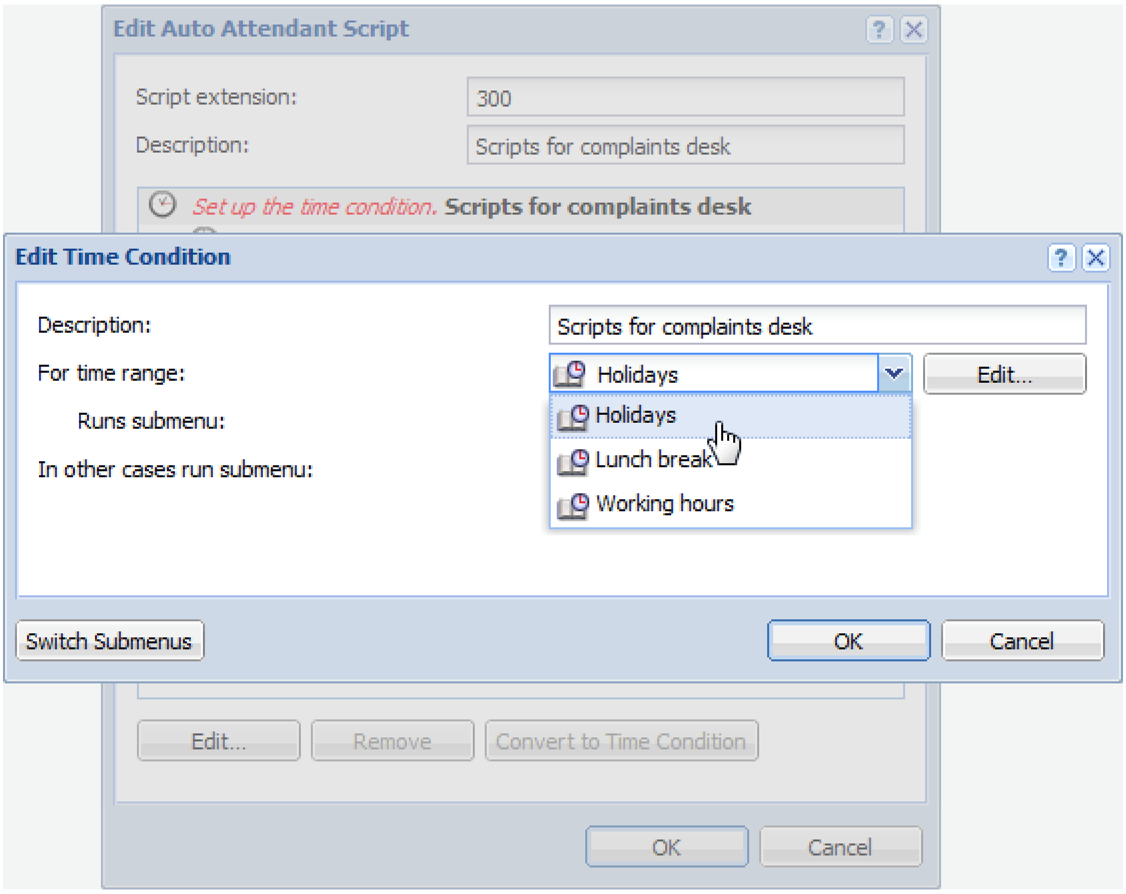
The script will follow this scheme:

```
If Holidays
    publicholidays.wav
Else
    If Working hours
        If Lunch break
           lunchbreak.wav
        Else
           Action: Redirecting to Alice’s extension.
    Else
        offtheclock.wav
```

1. In the administration interface, go to **Auto Attendant Scripts**.

2. Add a new script, assign it extension 300 and add a description (`Scripts for complaints desk`).

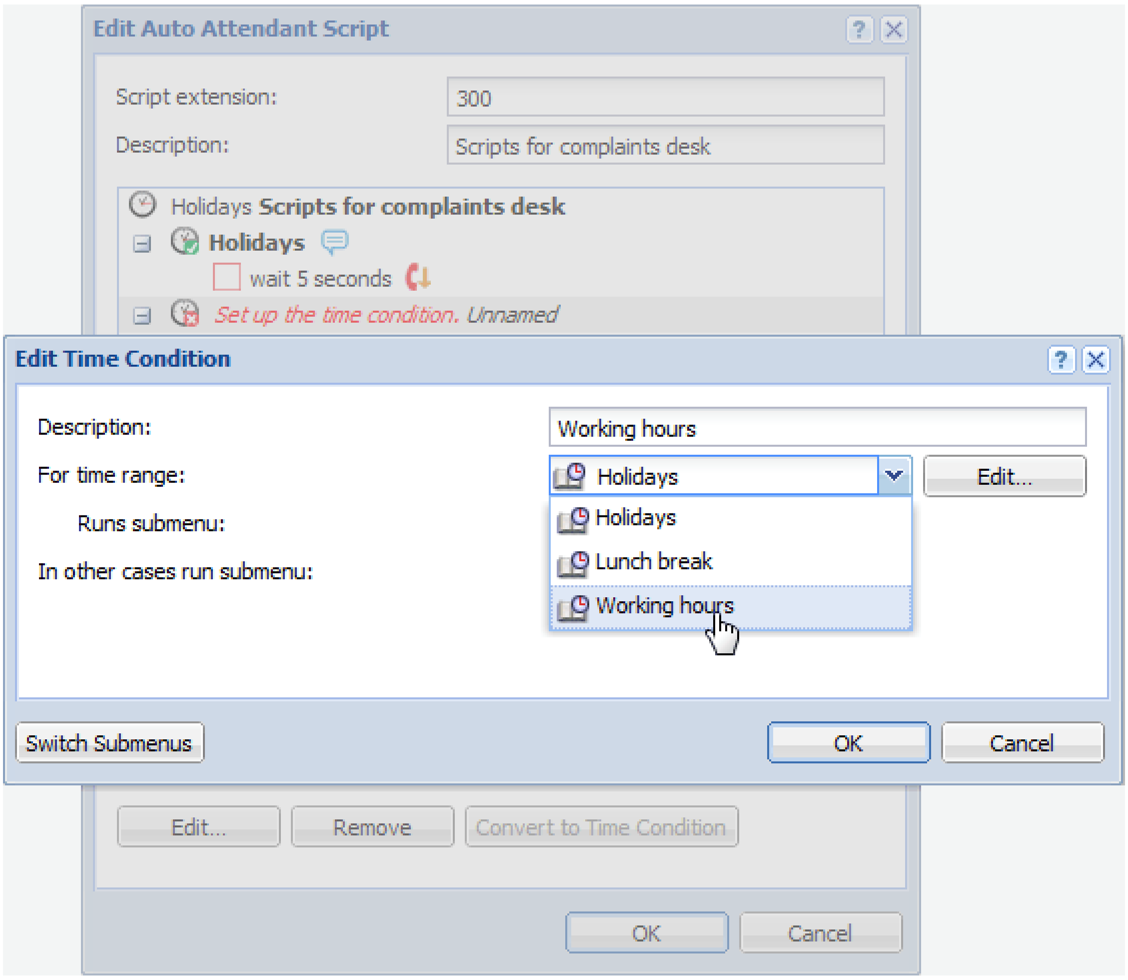
3. Create the first condition: Click **Convert to Time Condition**. Double-click on the red link **Set up the time condition** and in the **Edit Time Condition** dialog, select range `Holidays`. Save the settings.



4. Now, edit the first part of the condition in dialog **Add Auto Attendant Script**. Double-click *Unnamed*.

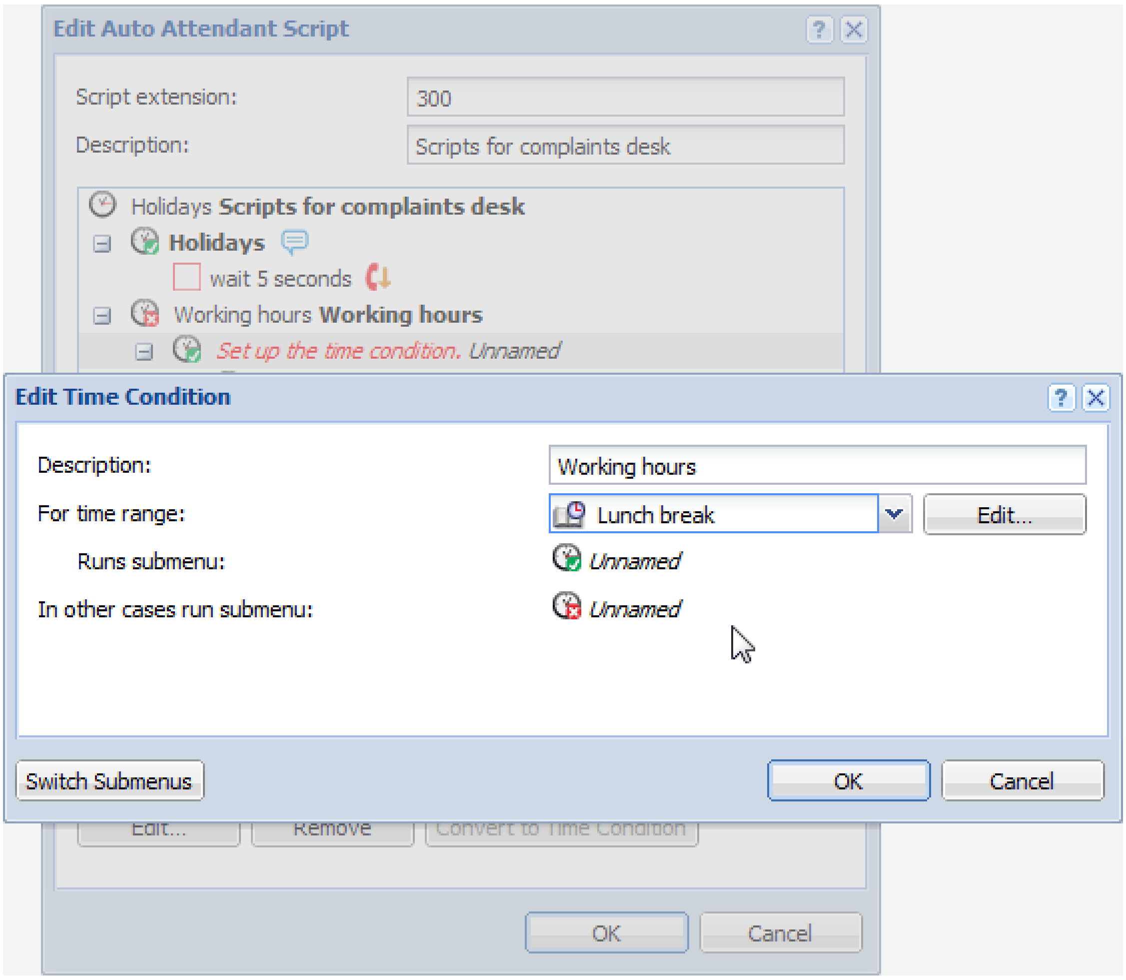
5. In the **Edit Menu** dialog, type description `Holidays` and add a file with the announcement about a holiday. Set timeout to 5 second (this will suffice) and save the settings.

6. Create the second condition: Select the *Unnamed* icon and click **Convert to Time Condition** (thus the Working hours condition will be nested into condition Holidays). In the **Description** field, enter **Working hours**; in the **For time range**, select **Working hours**. Save the settings.



7. In the **Edit Auto Attendant Script** dialog under the **Working hours** line, two new conditions appear.

8. Create the third condition: Click *Unnamed* and click **Convert to Time Condition**. In the **For time range** menu, select **Lunch break** Save the settings.



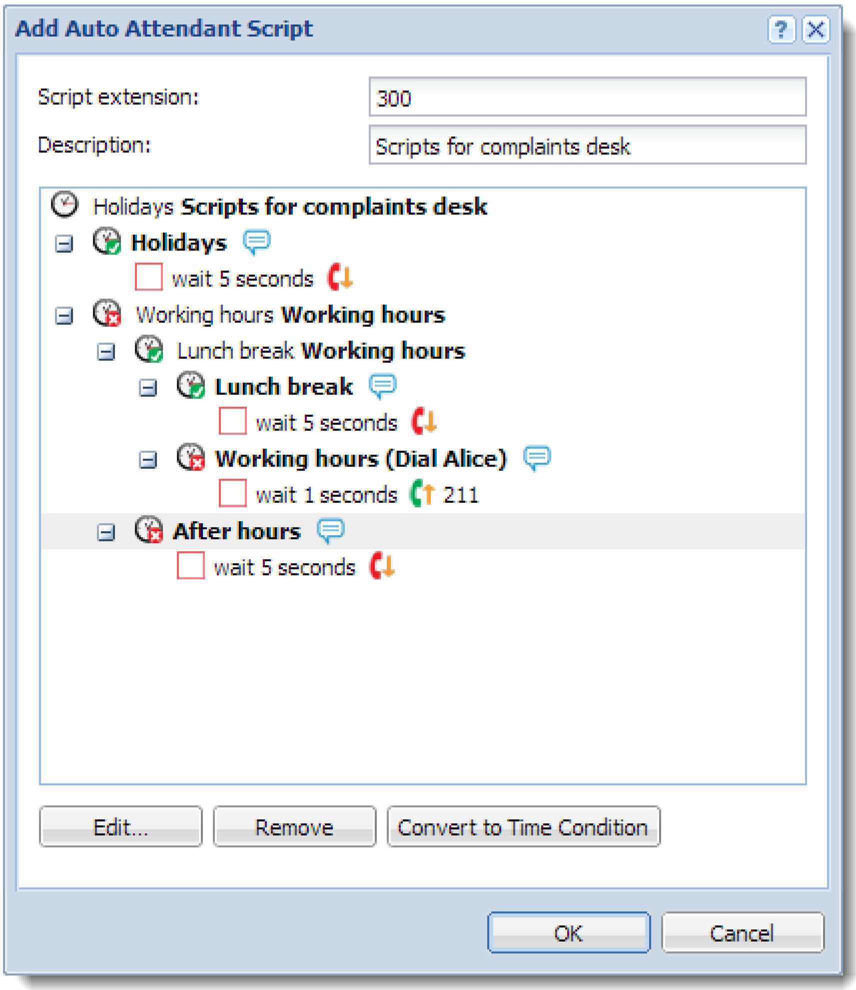
9. Double-click the last *Unnamed* icon. In the **Edit Menu** dialog, type description **Lunch break** and add a file with the announcement about a lunch break. Set timeout to 5 second and save the settings.

10. Double-click the *Unnamed* icon (last but one in the scheme). In the **Edit Menu** dialog, type description **Working hours (dial Alice)**. You can add an **Announcement** with information about redirecting to the Complaints department. Set **Timeout** to 1 second. In the **Default action** menu, select **Dial extension number**. Type Alice's extension (211) in the **Extension** field and save the settings.

11. Double-click the last condition (icon *Unnamed*). In the **Edit Menu** dialog, type description **After Hours** and add a file with the announcement that the Complaints department is close at the moment. Set timeout to 5 second and save the settings.

> **WARNING**
> If you open the administration interface in Safari browser and you cannot play any recordings, read topic Cannot play voicemails or audio files in Safari.

The resultant script is displayed below.



4.5.8 Using the Day/night mode in auto attendant scripts

> **NOTE**
> New in Kerio Operator 2.3.2!

Day/night mode works similarly to time ranges in auto attendant scripts. The difference is that in a time range you set a specific time. The Day/night mode is switched on demand.

An example:

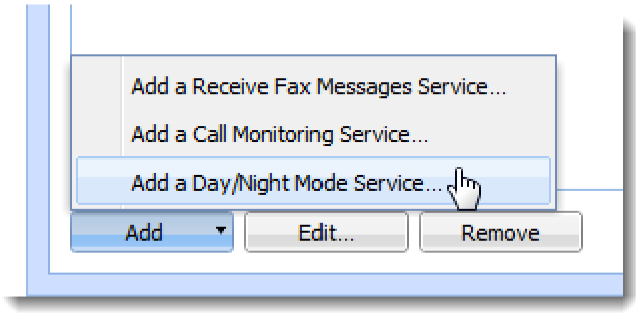
Alice works in a Sales department and has flexible working hours. She needs her extension to be available only when she's at work. She cannot use time ranges because her working hours are not fixed. She will use the Day/night mode to switch her extension on and off easily.

Alice must add a Day/night mode service to Kerio Operator and then create an auto attendant script using the created service.

Adding a Day/night mode

1. In the administration interface, go to section **Configuration > PBX Services**.

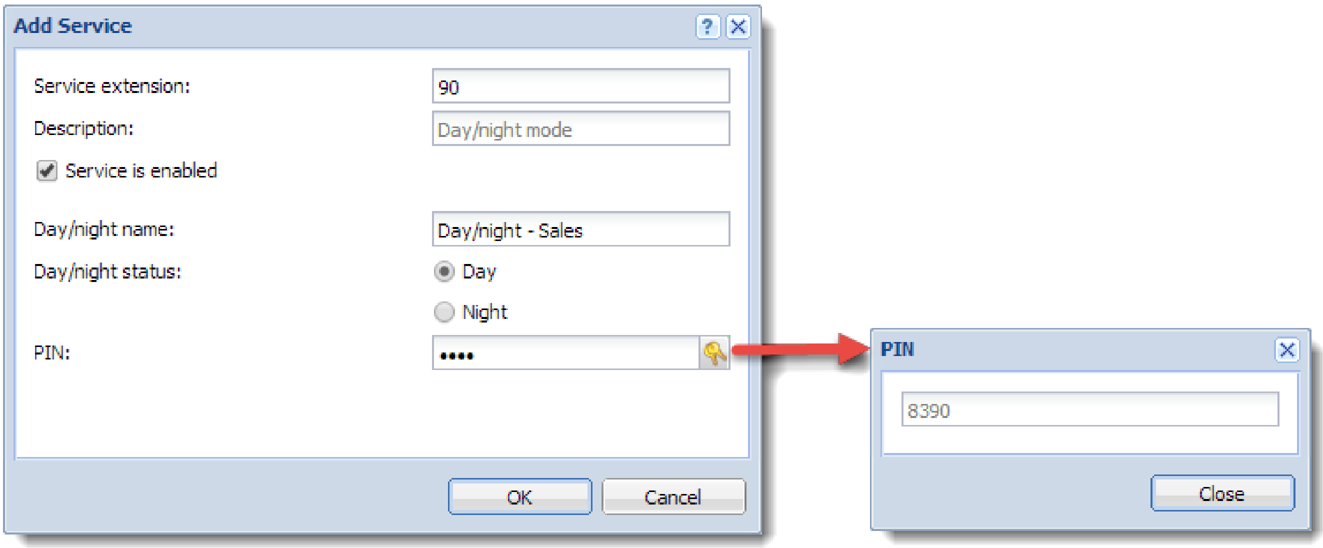
2. Click the **Add > Add a Day/night Mode Service**.



3. In the **Add Service** dialog window, fill in the service extension and type a name.

4. Click the two keys icon to display the service PIN number.

5. Click **OK**

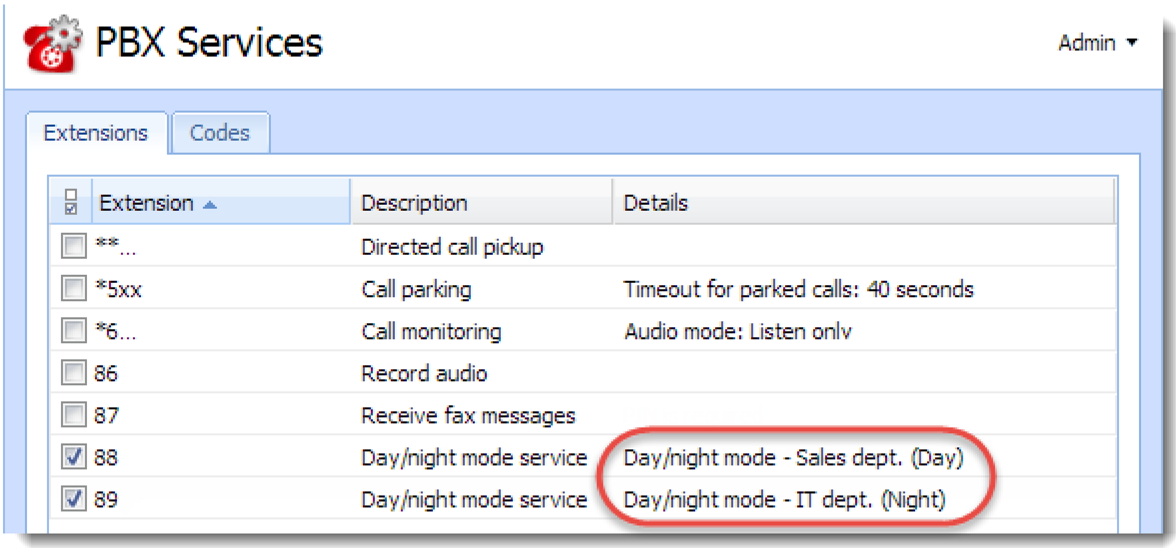


You can create as many Day/night mode services as you need, for example one for each department.

Selecting the Day/night mode status

You can select the mode status either manually in the administration interface or by calling the service extension.

The **Configuration > PBX Services** section displays the current status of each mode in the service list.

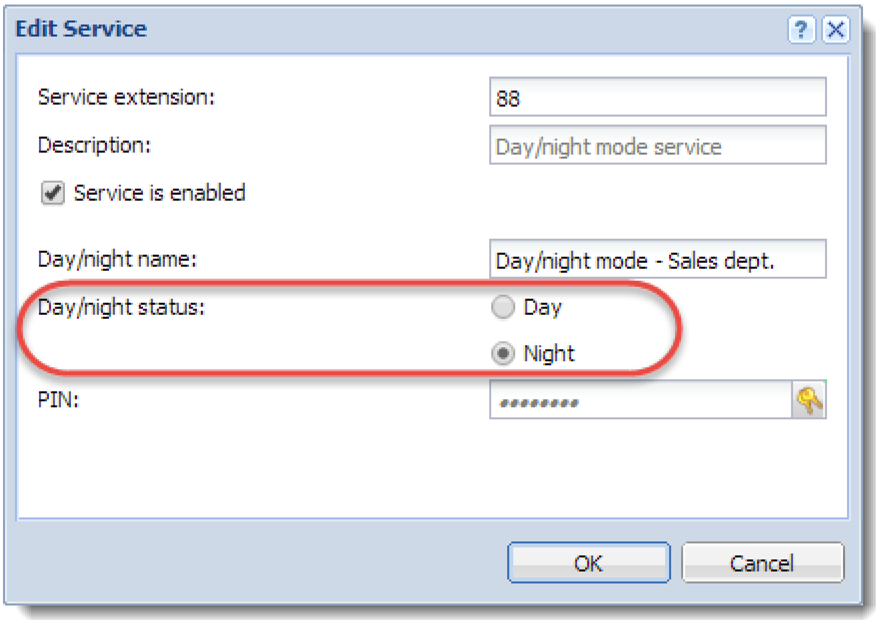


Selecting the status in the administration interface

1. In the administration interface, go to section **Configuration** > **PBX Services**.

2. Double-click the desired Day/night mode service.

3. Change the mode status.



4. Click **OK**

Selecting the status by calling the extension

1. Call the service extension (88 in our example).

2. Enter the service PIN code. When you switch to the day status, you hear a ascending sound. When you switch to the night status, you hear a descending sound.

3. Hang up.

Using day/night mode in auto attendant scripts

The script will follow this scheme:

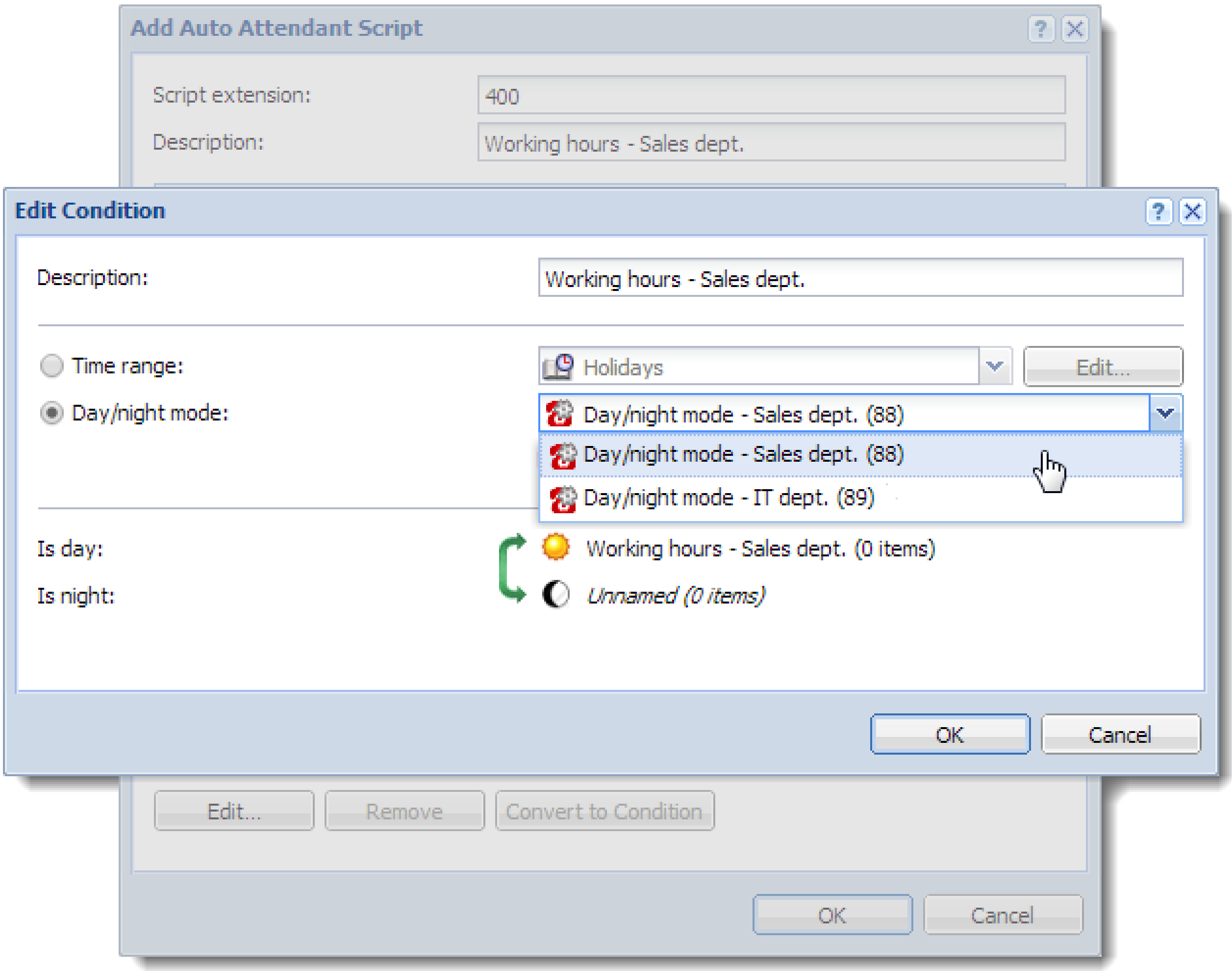
```
If Working hours
    Redirect to extension 256
Else
    Play offtheclock.wav
    Hangup
```

1. In the administration interface, go to **Auto Attendant Scripts**.

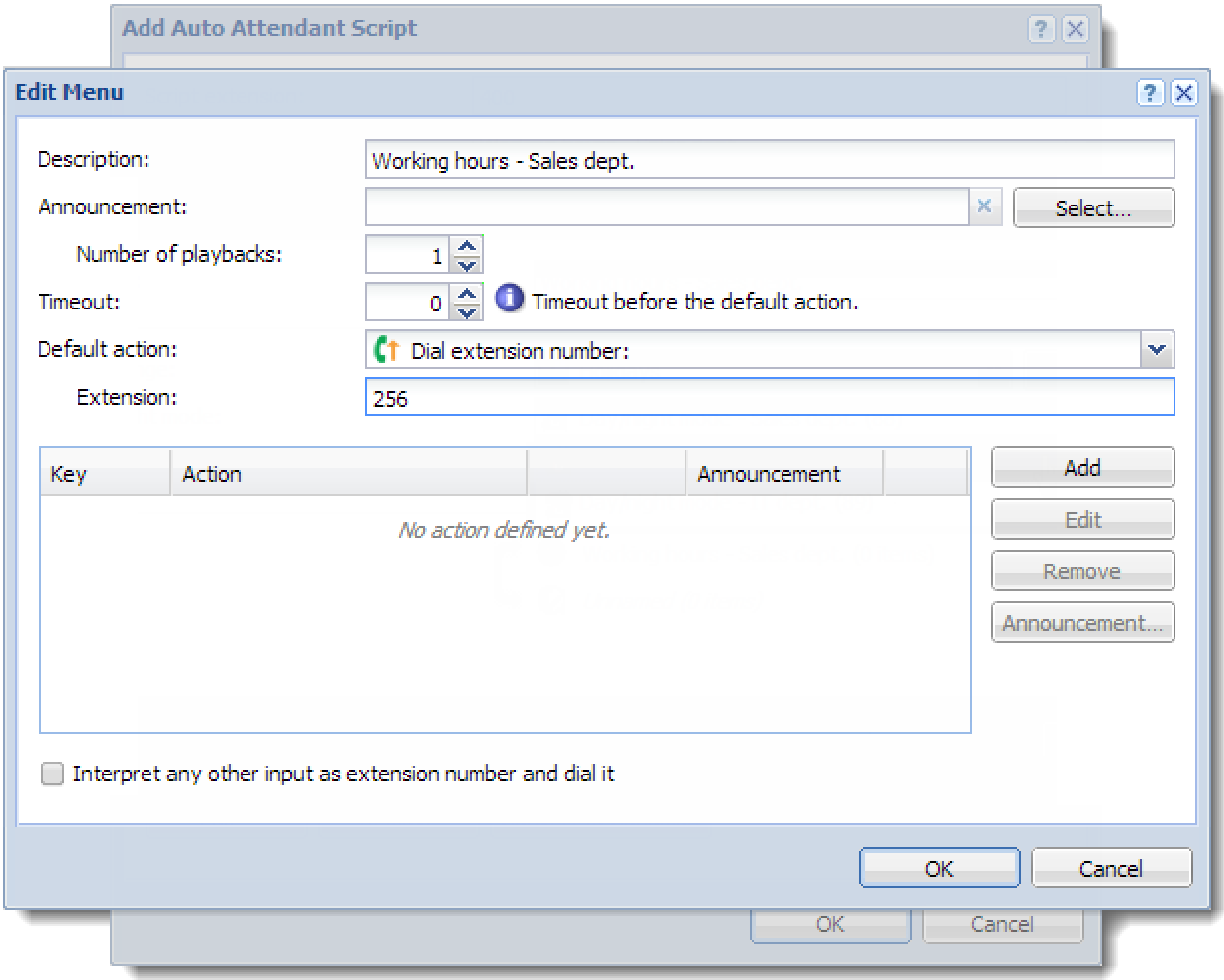
2. Add a new script, assign it extension 400 and add a description (`Working hours Sales dept.`).

3. Click the **Convert to Condition** button. The **Edit Condition** dialog opens.

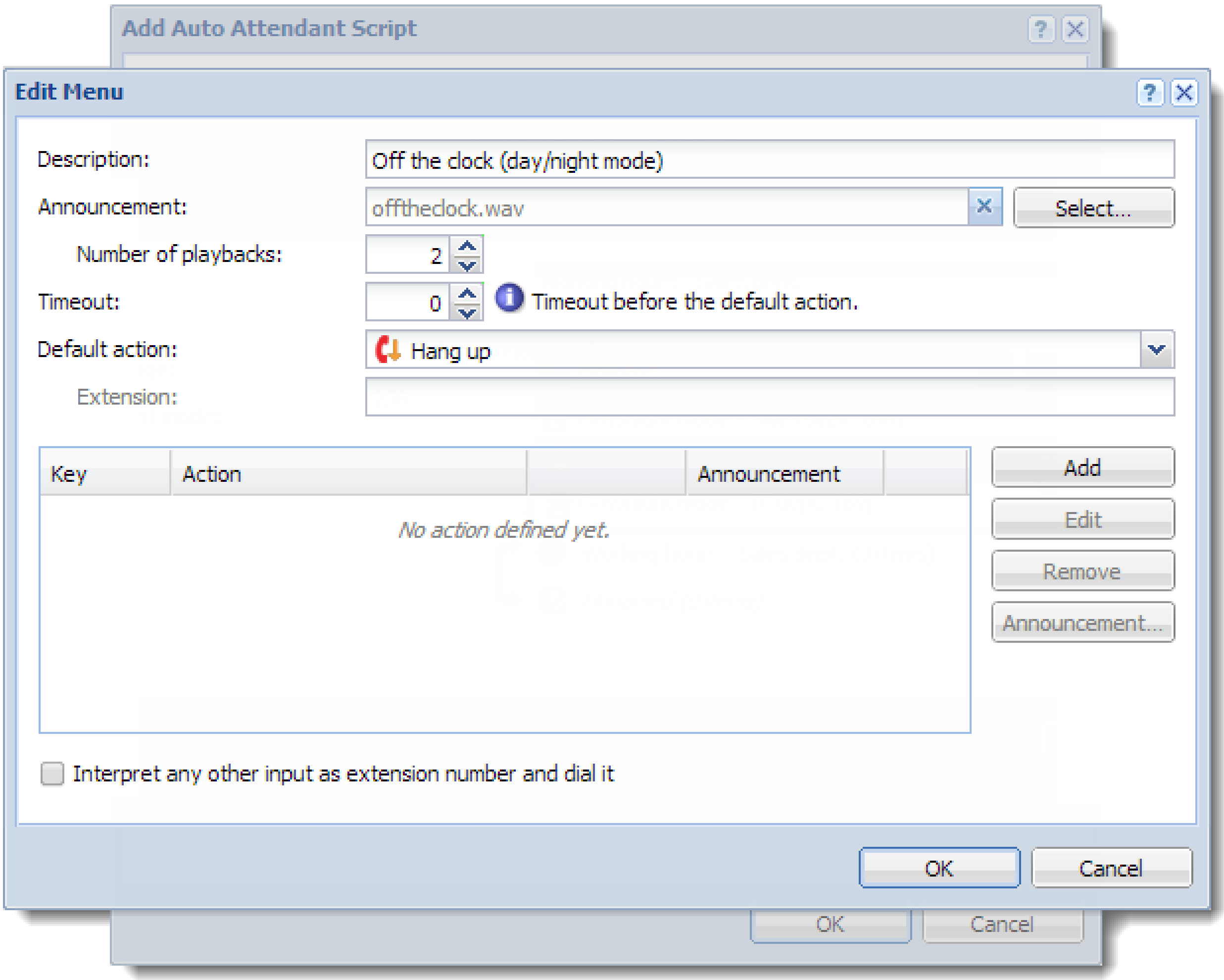
4. Select the **Day/night mode** mode from the drop-down menu. Click **OK**



5. Select the Day status (the sun icon) and click **Edit**. Set **Dial extension number** as the **Default action**. Type the extension number and click **OK**

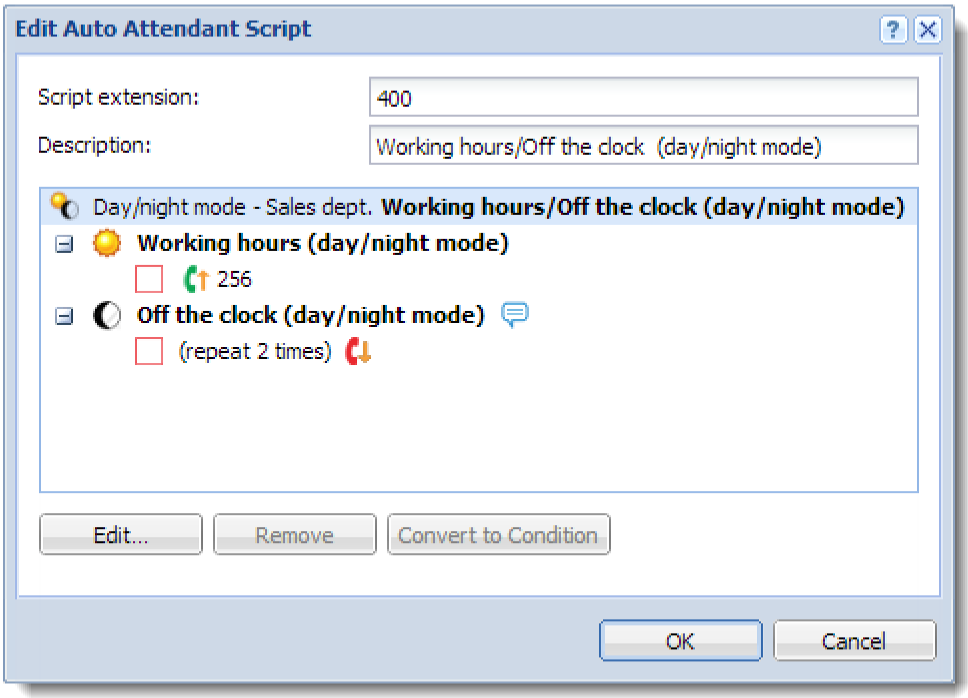


6. Select the Night status (the moon icon) and click **Edit**. Select the **Announcement** to be played when the night status is active (`offtheclock.wav` in our example). Set **Hang up** as the **Default action**. Click **OK**



> **WARNING**
> If you open the administration interface in Safari browser and you cannot play any recordings, read topic Cannot play voicemails or audio files in Safari.

The resultant script is displayed below.



Now Alice calls the Day/night mode extension (88 in our example) when she comes to work to switch the `day status` on. She calls the extension again to switch to the `night status` when she goes home.

4.5.9 Configuring call pickup

This function enables users to answer a call ringing on an extension on a device at another extension. The PBX distinguishes between two types of call pickup:

» Call pickup within defined groups (so called rooms) by using specific code (by default, this code is *8),

» Call pickup by using a special code (by default, this code is **) with the called extension appended at the end.

How to configure call pickup rooms

1. In the administration interface, go to **Configuration > PBX Services**, enable **Call Pickup** and save the settings. Keep the default pickup code (*8) unless you do have a reason to change it.

2. Go to **Definitions > Call Pickup Rooms** and click **Add** to open dialog **Add Call Pickup Room**.

3. Type the name of the department or the office in the **Name** field.

4. In the table, add all users and extensions that will be able to pick up calls for one another.

5. Make sure the **Room is enabled** option is checked.

> **Example**
>
> The Live And Let Laugh company network administrator uses the **Add Call Pickup Room** dialog to add a group with room name Local Sales for HPR (Happy people Republic). He adds all sales assistants for local market and their extensions: Frederic Jovial, Mary Merry, George Funpoker.
>
> Frederic Jovial has a day off today. His phone is ringing. Thanks to the call pickup rooms feature, Mary Merry does not need to dash for the Frederic's desk every time a customer calls his extension. She simply dials the magic code *8 and serves the customer at her desk.

How to configure directed call pickup

Directed call pickup is a service allowing to pickup calls directed to any extension at the PBX. Imagine the following situation:

- the managing director Peter Prank uses extension `101`
- the financial director Oscar Jape uses extension `102`
- they share an assistant, Ms Alessandra G. Uffaw.

If Alessandra's phone shows information that someone is calling the managing director (Peter Prank) during his meeting with the financial director (Oscar Jape), she can accept the call by dialing `**101`. Once she picks up the call, she learns that the caller is the International laughter Association manager and arranges a meeting for him and her company's executive manager. A few minutes later, the phone at the desk of the financial director Oscar Jape starts ringing. Again, the assistant can accept this call at her desk phone. now she enters the code `**102` and recommends the caller (the Cirque de Rire ringmaster) to call Mr Jape back later.

As you can see, by dialing the call pickup code, you can answer a call for any extension of the PBX.

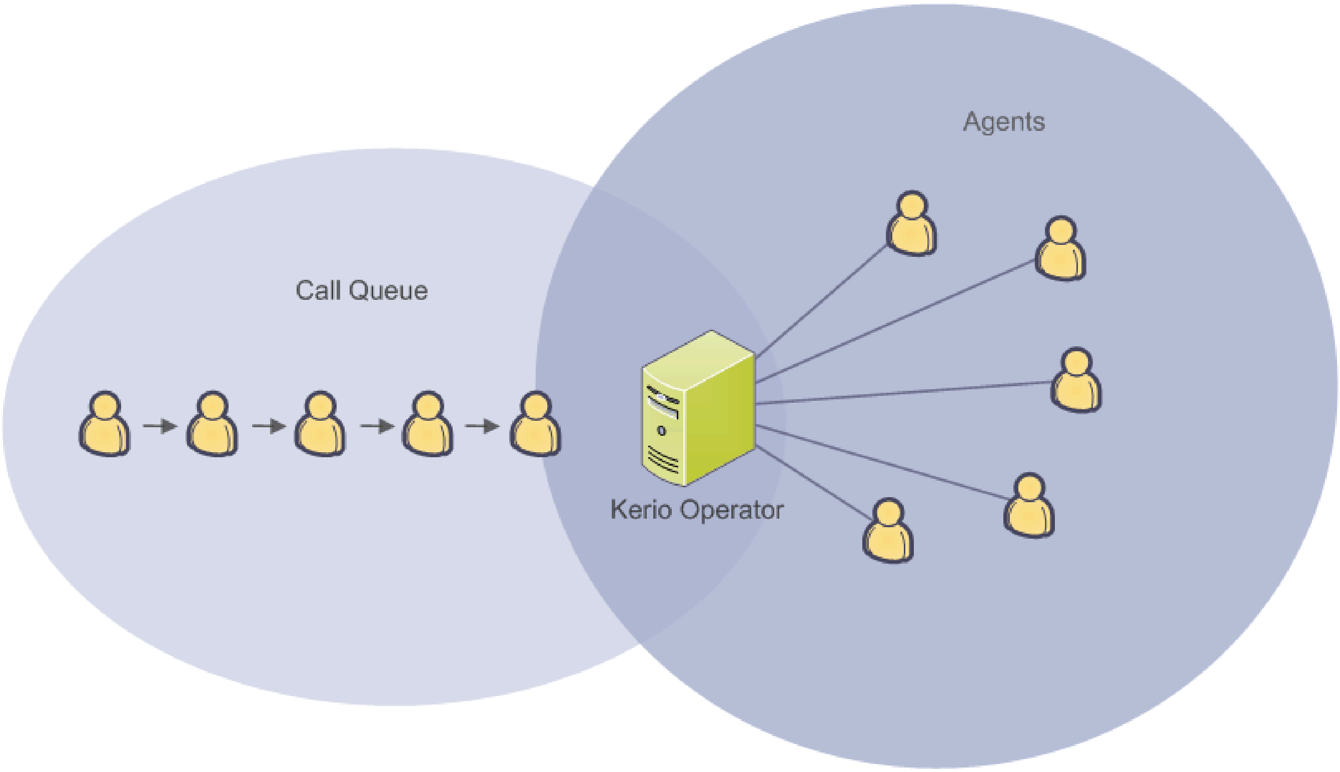
For directed call pickup, apply settings as described below:

1. In the administration interface, go to **Configuration > PBX Services**.
2. Enable **Directed Call Pickup**.
3. Directed call pickup is now fully functional.

> **NOTE**
> You can use directed call pickup in Kerio Phone.

4.5.10 Configuring call queues

Call queues are used to distribute incoming calls between agents.



Screenshot 52: Call queue

Configuring call queues

1. In the administration interface, go to **Configuration > Call Queues**.

2. Click **Add** to open the **Add Call Queue** dialog. On the **General** tab, type the new queue extension number.

3. Select the queue strategy.

4. Click the **Agents** tab.

5. If you want your agents to log in dynamically, type login and logout code. For example, 12345 to login, and 54321 to logout. The calls will only go to agents logged into the queue. If you want to assign specific agents permanently to the queue, click **Add** to select their extensions.

> **NOTE**
> Both methods can be combined. One queue may have agents who are assigned permanently and agents who log in dynamically.

6. Click the **Announcements** tab. An announcement is a pre-recorded message that callers hear while waiting in a call queue. You can import pre-recorded announcements into Kerio Operator (see topic Language settings in Kerio Operator) or record them by going to **Configuration > PBX Services > Record audio** (see topic Using PBX services).

How to select a queue strategy

» Round robin with memory mode uses circular call distribution. It remembers the last agent who answered the phone, and new calls are directed to the next agent in the round queue.

» Ring all agents — calls always ring at all agents until one of them answers the particular call.

» Ring least recently called agent — the system selects the agents who have not answered the phone for the longest period.

» Ring agent with fewest calls — the system assigns the call to the agent with the lowest number of calls answered so far.

» Ring random agent — if you select this option, the system will choose an agent randomly.

» Ring in order — only for permanently assigned agents. You specify a fixed order in which they are always selected. This strategy is for companies where all calls are answered by a receptionist. In case the receptionist is not answering, the call is directed to the next agent in order (for example, an administration assistant).

What is the difference between permanently assigned and dynamic agents

» Permanent assignment — agent's extension is assigned permanently to the queue.

» Dynamic login — agents use special code for logging in and out of the queue.

Recording calls from call queues

Kerio Operator allows recording calls from call queues. No other module or equipment is necessary. Setting can be done as follows:

1. Open the **Configuration > Call Queues** section and select the queue in which you wish to record the calls.

2. On the **General** tab, select **Record calls**.

To play back recorded calls

> **WARNING**
> Please note that call recording is subject to special laws in many countries. It may not be legal in your jurisdiction, or may require notice to the other party on the call. Accordingly, you assume all liability for using the call recording functions and are responsible for notifying all users of this system of this potential restriction, if applicable.

Section **Status > Recorded Calls** displays all calls recorded from call queues. Select a call to listen to it, download it to your computer or remove it.

Deleting Recorded Calls

Recorded calls can be periodically deleted once their total size reaches a certain limit. The limit can be set in section **Status > Recorded Calls**.

1. Click **Advanced > Periodically Remove Old Recorded Calls**.

2. In **Remove Old Recorded Calls** dialog box, enter the maximum size of recorded calls on a disk (in MB).

Configuring a call queue timeout

The call queue timeout period determines the maximum amount of time a caller can be placed in a call queue. Configuring the limit prevents from waiting in a queue infinitely.

The timeout limit is unlimited by default. For setting the limit, perform these steps:

1. In administration, go to **Configuration > Call Queues**.

2. Click **Add**/**Edit**.

3. On tab **General**, set **Queue timeout**.

4. (Optional) Go to tab **Announcements** and select **Timeout announcement**. Such announcement will play when the limit is reached and should include information about what happens next (tab Exceptions).

5. Go to tab **Exceptions**.

6. Choose an action for exceeded limit:

- **Callers receive a busy signal** — if announcement was set, recording plays before call termination.
- **Forward to** — type an extension. Kerio Operator forwards callers to the extension. If announcement was set, recording plays.

7. Save the settings.

Timeout is configured. If you want to check your settings, lower the limit to several seconds and dial the queue from several phones.

Configuring a music on hold and a while waiting period

A while waiting period is the period when users are waiting in a call queue for an agent. You can set what is playing during the period:

1. In administration, go to **Configuration > Call Queues**.

2. Click **Add**/**Edit**.

3. On tab **General**, select **While waiting**:

- Music on hold — a music sounds during the while waiting period.
- Ringtone — a ringtone sounds during the while waiting period.

4. If you selected the **Music on hold** option, select the particular recording in the **Music on hold** menu. If you want to add a new recording to Kerio Operator, go to the **Definitions > Music on Hold** section.

5. Save the settings.

Configuring a queue length

A queue length determines max. number of callers in the queue at the same time. Configuring the limit prevents from waiting too long in the queue.

The queue length is unlimited by default. For setting the limit, perform these steps:

1. In administration, go to **Configuration > Call Queues**.

2. Click **Add**/**Edit**.

3. On tab **General**, set **Queue length**.

4. (Optional) Go to tab **Announcements**, select **Full queue announcement**. Such announcement will play when the limit is reached and should include information about what happens next (tab Exceptions).

5. Click the **Exceptions** tab.

6. Select an action for exceeded limit:

- **Callers receive a busy signal** — if an announcement was set, recording plays before a call is terminated.
- **Forward to** — type an extension. Kerio Operator forwards callers to the extension. If an announcement is set, Kerio Operator plays the recording.

7. Save the settings.

The queue length is configured. If you want to check your settings, lower the limit to 1 and dial the queue from two phones.

Configuring exit keys

You can set exit keys for each call queue. Callers can use an exit key for transfer to an extension.

1. In administration, go to **Configuration > Call Queues**.

2. Click **Add**/**Edit**.

3. On tab **General**, click **Edit** next to **Exit keys**.

4. Edit **Exit Keys**, click **Add**.

5. In the **Add Exit Key** dialog, type an exit key (for example 1).

6. Type an existing extension to transfer calls.

7. Type a description.

8. Save the settings.

When users standing in the queue use the exit key, they are transferred to pre-configured extension.

Configuring call queues without agents

Follow these steps if no agents are logged into the queue:

1. In administration, go to **Configuration > Call Queues**.

2. Click **Add**/**Edit**.

3. (Optional) Go to the **Announcements** tab, select **No agents announcement**. Kerio Operator plays the announcement when there are no agents in the queue.

4. Go to tab **Exceptions**.

5. Select an action if the queue has no agents:

- **Callers cannot join the queue. Callers already waiting are removed** — Kerio Operator disconnects all callers. If **No agents announcement** is selected, Kerio Operator plays the recording.

- **Callers can join the queue** — new callers can connect to the queue. Current callers stay in the queue. If **No agents announcement** is selected, Kerio Operator plays the recording.

- **Callers cannot join the queue** — new callers cannot connect to the queue. Current callers stay in the queue. If **No agents announcement** is selected, Kerio Operator plays the recording.

6. If you selected **Callers cannot join the queue** or **Callers cannot join the queue. Callers already waiting are removed**, select one of these actions:

- **Callers receive a busy signal**

- **Forward to** — type an extension or external phone number. Kerio Operator forwards callers to the number.

7. Save the settings.

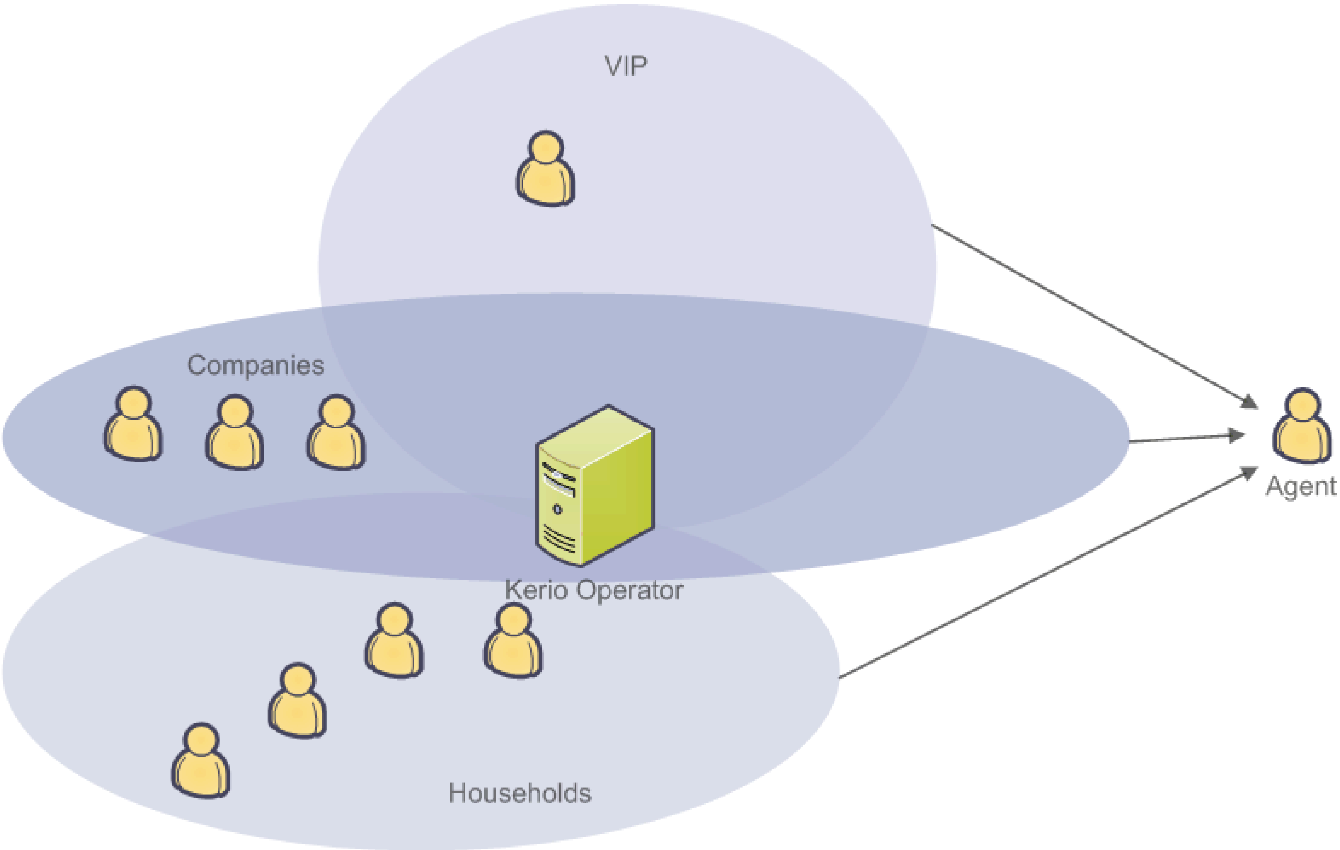
Settings are complete now. If you want to check your configuration, test these cases:

1. No agent serves the queue. Try to join the queue as a caller.

2. One agent serves the queue. Join the queue as a caller. Agent logs out.

Prioritizing call queues

Agents can operate several call queues. In the following example, an agent is assigned to three queues.



Screenshot 53: Operating multiple queues at once

To help agents identify the queues, you can upload various audio records for each queue. The record identifying the queue is played to the agent before a call from this queue is connected.

Upload new audio record as follows:

1. Select a call queue or create a new one in section **Configuration > Call Queues**.

2. In the displayed dialog, go to the **Announcements** tab.

3. Check the **Help agents identify the source queue by playing this announcement** and click on **Select**.

4. In the **Select Audio File** dialog box, double-click a record to select it, or upload your own record to Kerio Operator (it must be in WAV or GSM format). Use the **Upload** button.

It is also possible to set priorities for individual queues:

1. Open the **Configuration > Call Queues** section.

2. Select a queue or create a new one.

3. In the displayed dialog, go to the **General** tab and set the desired priority.

4. Repeat the configuration for other queues.

Queues with higher priority are processed first.

Displaying missed calls on phones in call queues

When an incoming calls rings in the call queue and an agent answers it, other devices in the queue register the call as missed anyway.

To not display missed calls on other devices:

1. In the Kerio Operator administration interface, go to **Configuration > Call Queues**.

2. Select an extension and click **Edit**.

3. Switch to **Advanced** tab.

4. Select **Do not display missed calls on the phones**.

5. Click **OK**.

Monitoring active call queues

1. In the administration interface, go to section **Status > Call Queues**.

2. The top table shows currently active queues.

3. The other tables display agents and callers in a queue. Just select a queue and the details in table **Agents** and **Callers** are updated.

You can also reset the call queue statistics to start from zero. Use the **Reset Statistics** button.

4.6 Security

This section helps you secure your Kerio Operator server.

4.6.1 Securing Kerio Operator

Key measures

You can take the following measures to secure Kerio Operator:

» Restrict communication on firewall to necessary IP addresses and ports, especially if the PBX runs in the Internet.

» Restrict communication on the integrated firewall in Kerio Operator.

» Create strong SIP passwords.

» Restrict the number of attempts to enter SIP passwords.

» Using special rules, forbid international outgoing calls to countries you do not communicate with

» Restrict international outgoing calls to countries where you rarely call

» Encrypt your calls

» Encrypt data

The following sections describe these settings in detail.

Configuring firewall in local network

Kerio Operator is usually protected by firewall (in your local network or in the Internet). Certain ports need to be opened (or mapped) on firewall.

Service (default port)	Outbound connection	Inbound connection
SIP (5060)	allow	allow for SIP servers of your provider
IMAP (143)	allow if integration with Kerio Connect is enabled and there is a firewall between Kerio Connect and Kerio Operator.	deny
LDAP (389)	allow	deny
LDAPS (636)	allow	allow if you use mapping from Active Directory or Open Directory and there is a firewall between the directory service and Kerio Operator.
HTTP (80)	allow	deny
HTTPS (443)	allow	allow if you wish users to be able to connect to Kerio Phone from the Internet.
HTTPS (4021)	allow	allow if you wish users to be able to connect to the administration interface from the Internet.
STUN/TURN (3478)	allow	allow
STUN/TURN (3479)	allow	allow

Configuring firewall integrated in Kerio Operator

Prepare groups of IP addresses which you wish to allow for individual services (create them in **Definitions > IP Address Groups**).

You can configure the integrated firewall in section **Network > Firewall**.

Service	Recommendation
Web server	If you want to restrict connections to Kerio Operator administration and softphone, check this option and select an IP group with addresses from which access will be allowed. Bear in mind that all the PBX users should be allowed to connect to Kerio Phone at least from their own workstation.
SIP	We recommend to restrict the SIP protocol solely to your internal network and external IP addresses of your SIP provider.
Phone provisioning	For security reasons, we recommend to restrict automatic phone provisioning solely to your internal network because TFTP sends configuration data as plain text.
CRM integration	For security reasons, we recommend to restrict communication solely to your internal network.
SNMP monitoring	For security reasons, we recommend to restrict communication solely to your internal network and IP adresess where monitoring servers are running.

> **NOTE**
> If the options are unchecked, no restrictions are set.

Data Encryption

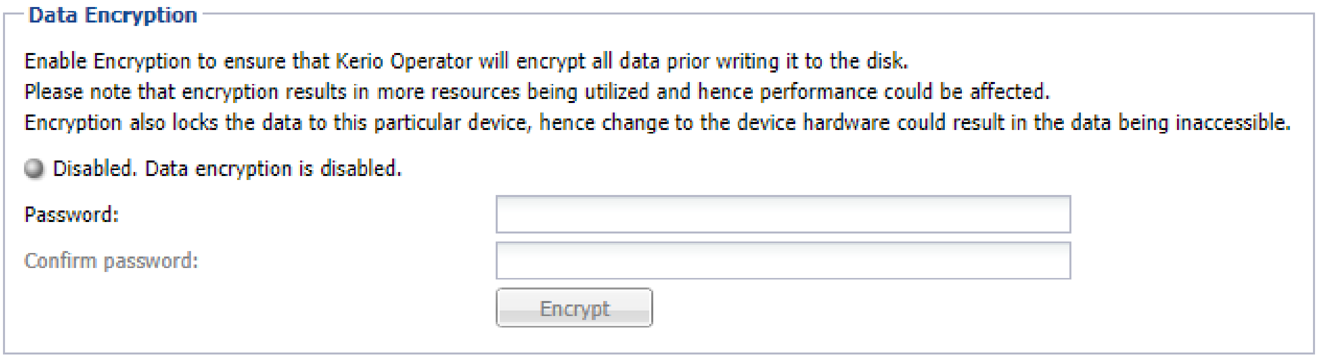
You can enable encryption to ensure that Kerio Operator encrypts recorded calls, voicemail messages, logs, and configuration before writing it to the disk.

> **IMPORTANT**
> Encryption is bound to a specific storage device, so if you plan to change the hardware you must first disable encryption. Also, encryption results in more resources being utilized so performance maybe impacted.

Enabling Encryption

1. In the Kerio Operator administration interface, go to **Configuration > Advanced Options**.

2. Go to the **Data Encryption** tab.



Screenshot 54: The data encryption tab

3. Key-in the **Password** and re-enter to confirm the same.

> **IMPORTANT**
> Once encryption is enabled, the password cannot be changed. Remember this password, as you would require it to decrypt data.

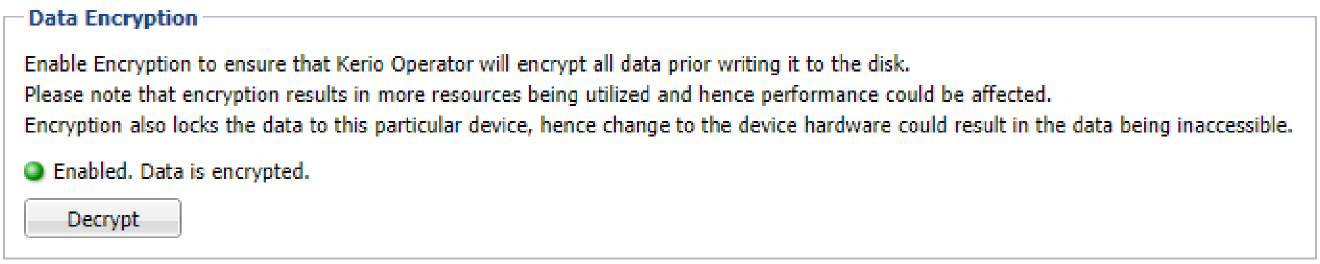
4. Click **Encrypt** and confirm the action.

Disabling Encryption

To decrypt: data and disable encryption:

1. In the Kerio Operator administration interface, go to **Configuration > Advanced Options**.

2. Go to the **Data Encryption** tab.



Screenshot 55: The data encryption tab

3. Click **Decrypt**.

4. Key-in the **Password** set while encrypting and confirm the action.

Configuring protection against password guesses

Login data guess is one of the most common attacks on a PBX. In Kerio Operator, attackers try to guess extension numbers and SIP passwords. This type of attack is defined by many unsuccessful attempts to enter extension number and SIP password during a login. Kerio Operator security settings enable you to limit the number of attempts of a phone (both software and hardware) to connect to the PBX. Apply settings as described below:

1. In the administration interface, go to **Security**.

2. Set the limit of unsuccessful attempts (usually 3 to 10 attempts) and set the time period during which attempts will be counted. Setting the time period protects real users who have forgotten their password or who have made mistakes during several logins. When the time limit expires, they can try to login to the PBX again.

3. Set the time during which Kerio Operator will block the source IP address.

4. You can also enter an email address that will be used for sending warnings about blocked IP addresses.

How to recognize there has been an attack attempt

In log **Security** look for the `Authentication failed` string. If there are many messages of this kind, somebody is trying to use a dictionary attack.

What to do in case of an attack

In case of an attack, apply the following instructions as soon as possible:

1. In section **Status > Calls** and in logs, look for information on which account has been abused.

2. Change the SIP password of this account.

3. Instruct users about handling their login details and secure behavior on the Internet.

4. The PBX is blocked, so it needs to be unlocked again.

4.6.2 Configuring SSL certificates

To secure the PBX by SSL/TLS encryption, you need a SSL certificate. SSL certificates authenticate an identity on a server.

Kerio Operator creates the first self-signed certificate during the installation. The server can use this certificate but users will have to confirm they want to go to an untrustworthy page. To avoid this, generate a new certificate request in Kerio Operator and send it to a certification authority for authentication.

> **WARNING**
> If you use the Safari browser in your environment (on Apple iPhone, Apple iPad, Mac OS X systems and on Microsoft Windows), you will not be able to play voice messages in Kerio Phone on their devices with a self-signed certificate. You must have a trustworthy certificate available.
> If you use a self-signed certificate, users with Apple mobile devices will not be able to play voice messages in Kerio Phone on their devices. They must have a trustworthy certificate available.
> To encrypt the communication between Kerio Operator and hardware phones (and only a self-signed certificate available), you have to import or configure information in the phones that the invalid certificate is to be ignored.

Creating self-signed certificates

To create a self-signed certificate, follow these instructions:

1. In the Kerio Operator administration interface, open section **Definitions > SSL Certificates**.

2. Click **New > New Certificate**.

3. In the **New Certificate** dialog box, type the hostname of Kerio Operator, the official name of your company, city and country where your company resides and the period of validity. The **Hostname** and **Country** entries are required fields.

4. Click **OK**.

5. To enable the server to use this certificate, select the certificate and click **Set as Active**.

Creating certificates signed by certification authority

If you wish to create and use a certificate signed by a trustworthy certification authority, follow these instructions:

1. In the Kerio Operator administration interface, open section **Definitions > SSL Certificates**.

2. Click **New > New Certificate Request**.

3. In the **New Certificate Request** dialog box, type the hostname of Kerio Operator, the official name of your company, city and country where your company resides and the period of validity. The **Hostname** and **Country** entries are required fields.

4. Click **OK**.

5. Select the certificate and click **Export**.

6. Save the certificate to your disk and email it to a certification organization.

7. Once you obtain your certificate signed by a certification authority, go to **Definitions > SSL Certificates**.

8. Click **Import**.

9. To enable the server to use this certificate, select the certificate and click **Set as Active**.

Intermediate certificates

Kerio Operator supports **intermediate** certificates.

To add an intermediate certificate to Kerio Operator, follow these steps:

1. In a text editor, open the server certificate and the intermediate certificate.

2. Copy the intermediate certificate into the server certificate file and save. The file may look like this:

```
-----BEGIN CERTIFICATE-----
MIIDOjCCAqOgAwIBAgIDPmR/MA0GCSqGSIb3DQEBBAUAMFMxCzAJBgNVBAYTAl
MSUwIwYDVQQKExxUaGF3dGUgQ29uc3VsdGluZyAoUHR5KSBMdGQuMR0wGwYDVQ
     ..... this is a server SSL certificate ...
ukrkDt4cgQxE6JSEprDiP+nShuh9uk4aUCKMg/g3VgEMulkROzFl6zinDg5grz
QspOQTEYoqrc3H4Bwt8=
-----END CERTIFICATE-----
-----BEGIN CERTIFICATE-----
MIIDMzCCApygAwIBAgIEMAAAATANBgkqhkiG9w0BAQUFADCBxDELMAkGA1UEBh
WkExFTATBgNVBAgTDFdlc3Rlcm4gQ2FwZTESMBAGA1UEBxMJQ2FwZSBUb3duMR
     ..... this is an intermediate SSL certificate which
           signed the server certificate...
5BjLqgQRk82bFi1uoG9bNm+E6o3tiUEDywrgrVX60CjbW1+y0CdMaq7dlpszRB
```

```
t14EmBxKYw==
-----END CERTIFICATE-----
```

3. In the administration interface, go to section **Definitions > SSL Certificates**.

4. Import the modified server certificate by clicking on **Import > Import a New Certificate**.

> **NOTE**
> If you have multiple intermediate certificates, add them one by one to the server certificate file.

Securing Kerio Phone with SSL certificates

For more information about securing Kerio Phone, see the Securing Kerio Phone with SSL certificates section in the **Provisioning for Kerio Phone** topic.

4.6.3 Configuring NAT

Kerio Operator is behind NAT and phones are on the Internet

1. In the administration interface, open section **Network > General**.

2. In the NAT support section, enable NAT by checking the option.

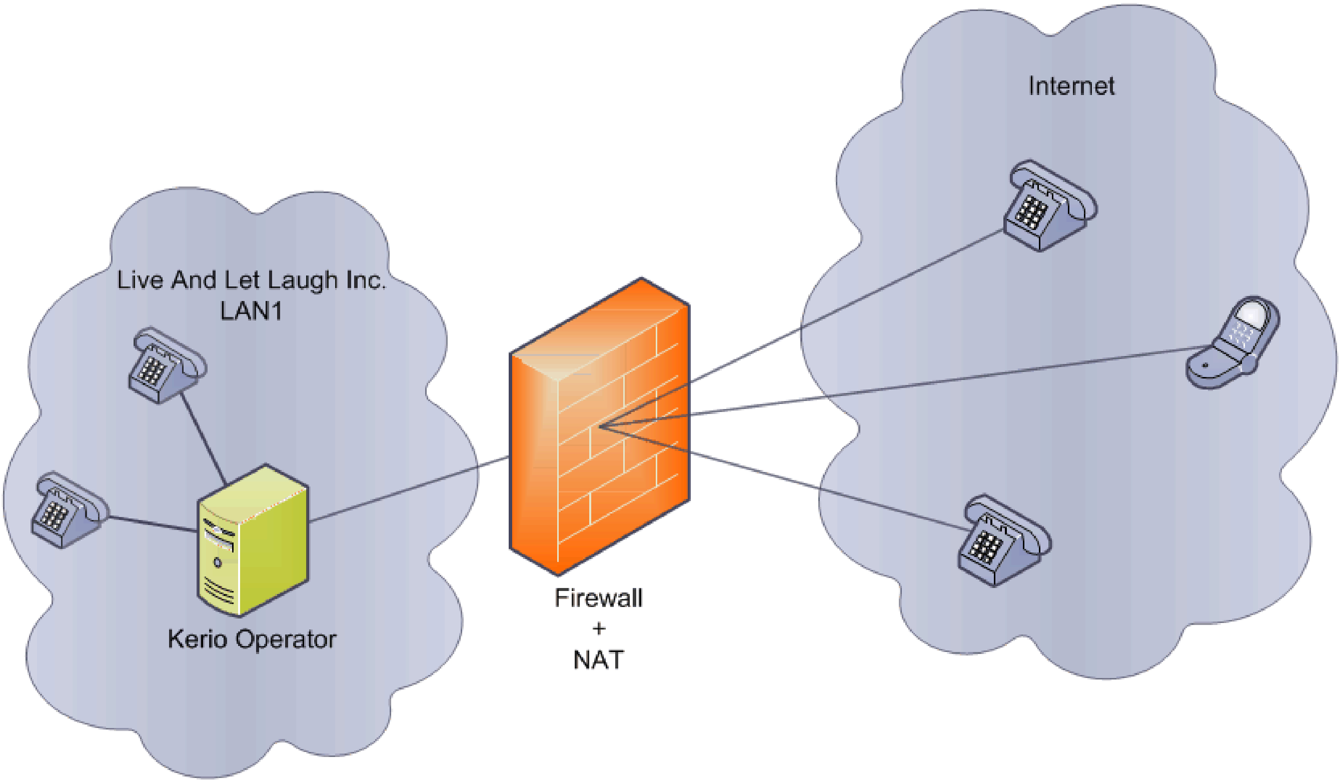
3. Enter the public address which should be used in SIP protocol messages.

4. For phones in the same private network as Kerio Operator, create an appropriate IP address group in section **Configuration > Definitions > IP Address Groups** with all addresses on which phones communicate in your private network. Thus, the PBX will communicate with phones within the network directly.

5. (Optional) You can also limit the RTP port range. Each call requires 4 ports for communication.

6. Also, map the following ports from firewall to Kerio Operator. It is usually necessary to map a port range for RTP (according to the specified interval).

- TCP+UDP/5060
- TCP/5061
- UDP/443
- TCP+UDP/3478
- TCP+UDP/3479



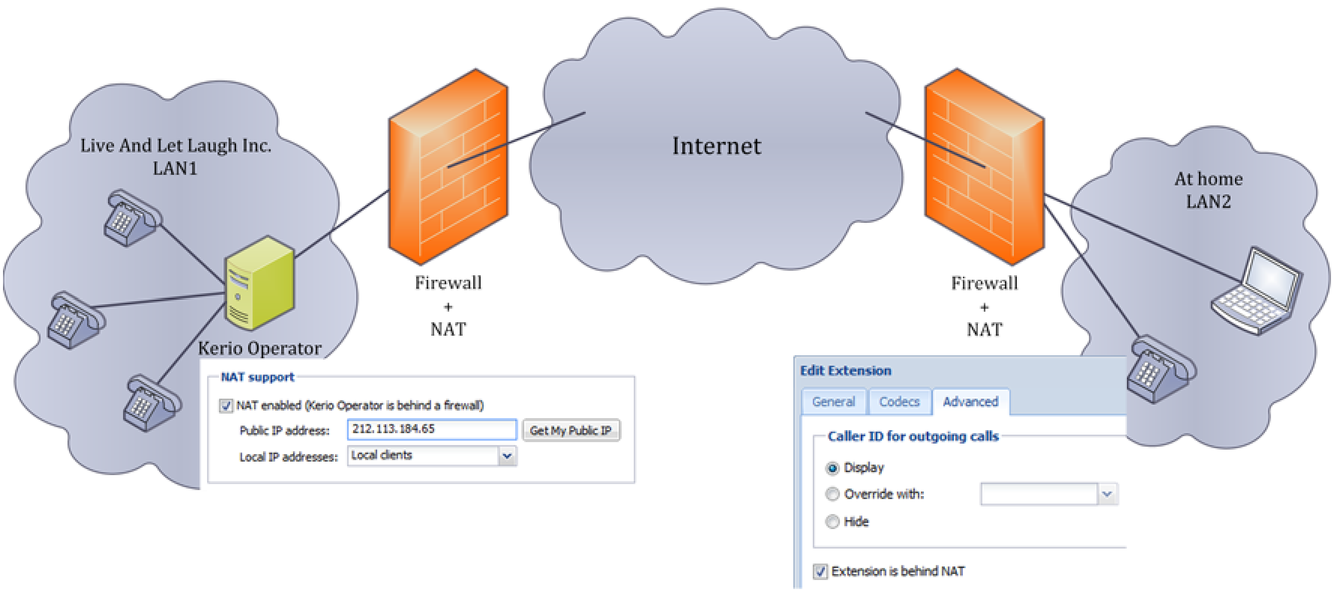
Screenshot 56: Kerio Operator is behind NAT and hardware phones are in the Internet

Kerio Operator is on the company network and hardware phones are behind NAT

Firstly, configure NAT for Kerio Operator.

The scenario in figure bellow requires only one minor configuration in the PBX settings:

1. In the administration interface, open the **Extensions** section.
2. Select the extension of the user whose phone is in a private network.
3. In the **Edit extension** dialog, go to tab **Advanced**.
4. Check the **Extension is behind NAT** option.



Screenshot 57: Kerio Operator is in the company network and hardware phones are behind NAT

Kerio Operator is behind NAT and hardware phones are on the Internet

Firstly, configure NAT for Kerio Operator.

If the telephone is in the Internet (not behind NAT), Kerio Operator does not require special configuration.

> **WARNING**
> Phones which are in the Internet cannot be managed in section Provisioned Phones.

4.7 Server settings

This section contains information about:

4.7.1 Language settings in Kerio Operator

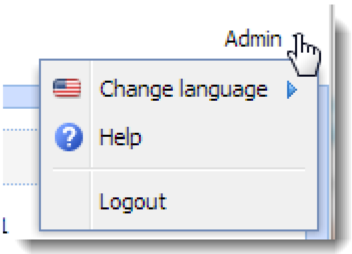
Languages in Kerio Operator are:

- Application language — language for the administration interface and for Kerio Phone.
- PBX language — the voice of the PBX. Voice records which are used for communication with users (internal and external).

You can also change the type of indication tones according to individual countries (read section Changing indication tones according to countries).

Changing the application language

The language for the administration and softphone interfaces can be set in the **Admin** menu in the right top corner of the of the application window.



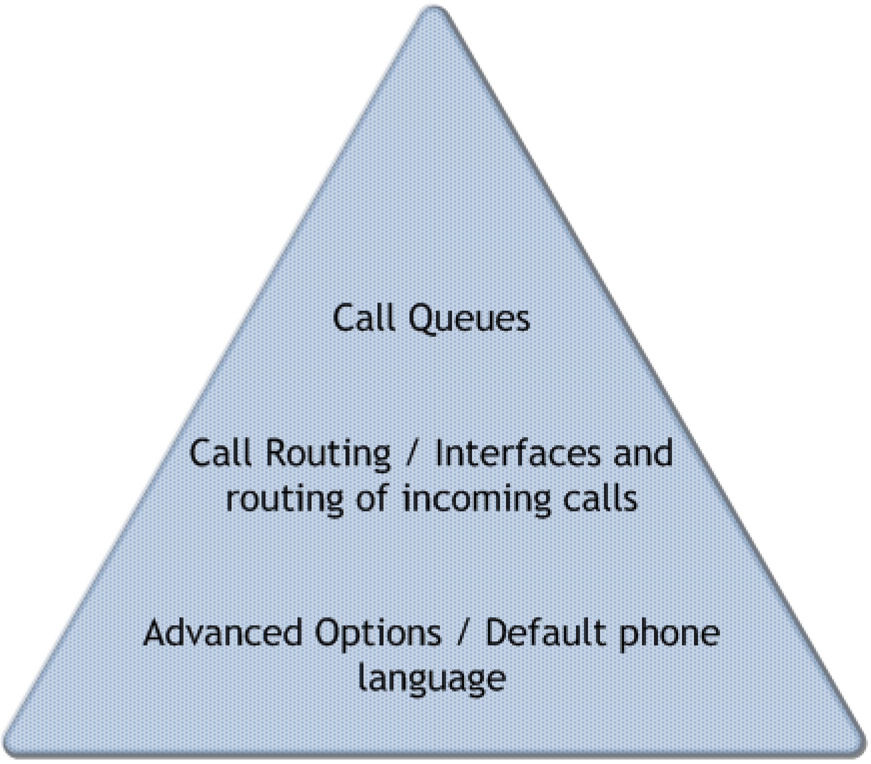
Changing the language of the PBX

You can change the default language of the PBX in the administration interface in section **Configuration > Advanced Options > Telephony**.

There, you can also upload new language version or different voice records of the same language (for example, less formal records).

When setting language, bear in mind the following rules:

- Default language set in section **Advanced Options > Telephony** has lower priority than settings of individual users in section **Users**. If users do not have any language set, the default one is used.
- Default language set in section **Advanced Options > Telephony** has lower priority than settings for interfaces for incoming calls (section **Call Routing**). The language set for the interface of incoming calls has lower priority than files uploaded to call queues (see screenshot below). If no language is set, the default one is used. The same goes for call queues.



How to change the language for individual users

Thomas Punchline, the network administrator at Live And Let Laugh Inc, faces the following problem: New employee has arrived in the company. Alessandra G. Uffaw has moved from the Bliss Seekers Land to the Happy People Republic and cannot speak the Happish language. She complains she can't understand her voicemail. Thomas has to switch the PBX language to the Cravish language for her. Do you need to solve a similar problem? Check the following example:

1. In the administration interface, go to **Configuration > Users**.

2. In the user's settings, go to tab **General** and change the **Phone language**.

How to change the PBX language for a group of users

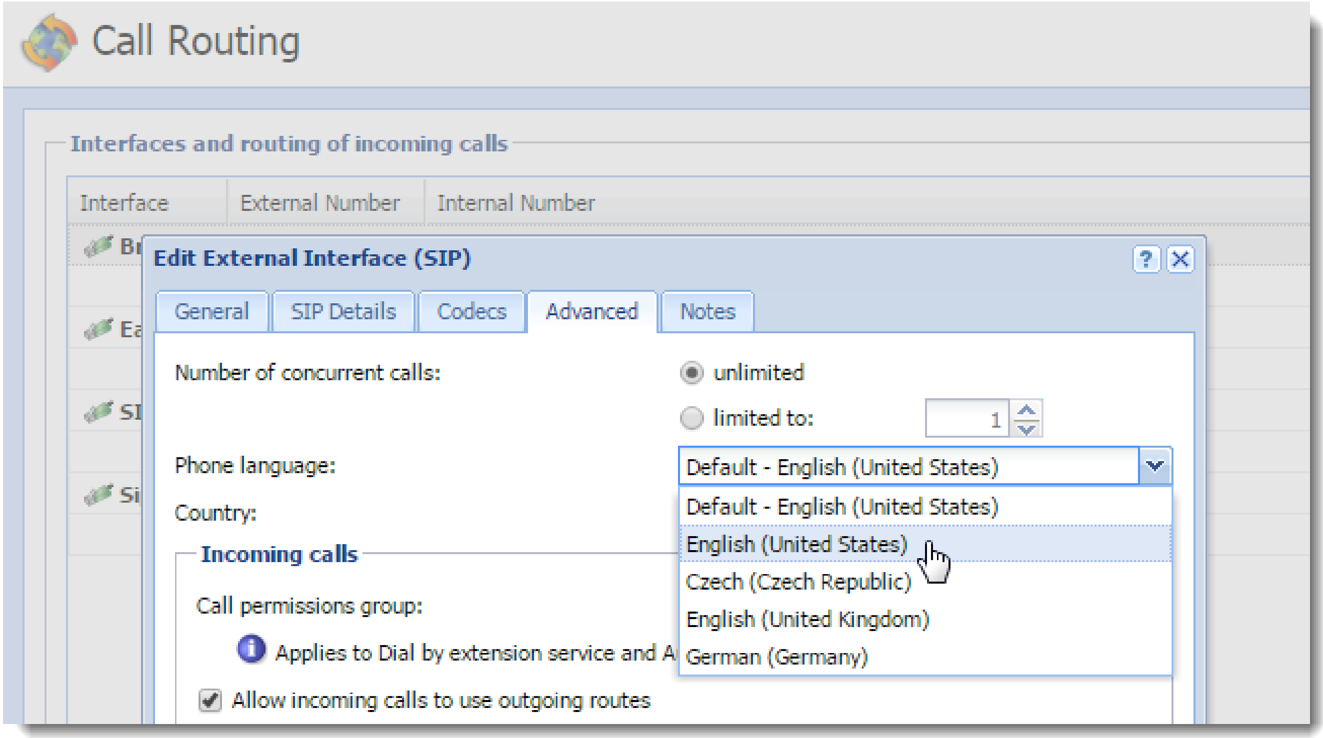
Thomas was instructed to create a new interface in Kerio Operator and change its language to the Cravish. He has to create a new interface for incoming calls and set a language for this interface. He called his VoIP service provider and purchased new phone numbers for the employees who will communicate with foreign customers. And how he configured Kerio Operator?

1. In the administration interface in section **Configuration > Call Routing** , add a new route for incoming calls.

2. Connect it to the provider, open the edit dialog by clicking on the route in table **Interfaces and routing of incoming calls**.

3. Select a language on tab **Advanced**.

4. Select a country on tab **Advanced**. Each country has different standards for indication tones during calls (e.g. beeps, ringing tones, etc.).



Setting a different language for a call queue

If you wish to change the language for call queues, not for the entire route, go to section **Configuration > Call Queues**.

> **NOTE**
> Language files used in call queues has automatically higher priority than language set for incoming calls.

How to add a new language to the PBX

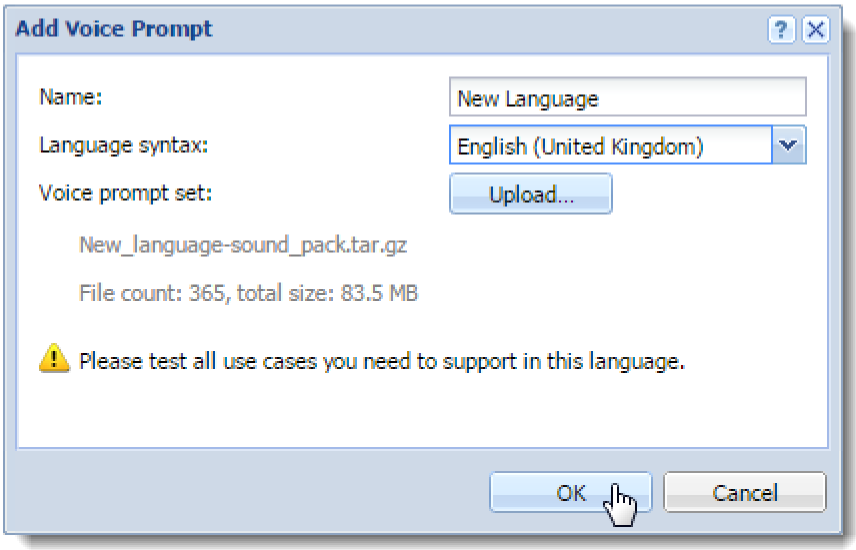
If the language sets (voice records) provided in Kerio Operator do not satisfy your needs, you can download or buy different language sets and import them to the PBX. You can download the language sets (free or paid), for example, in the following sites:

- http://www.voip-info.org/
- http://downloads.asterisk.org/

You can extract any language set archive and create your own voice records (provided you keep the file structure).

To add a new language:

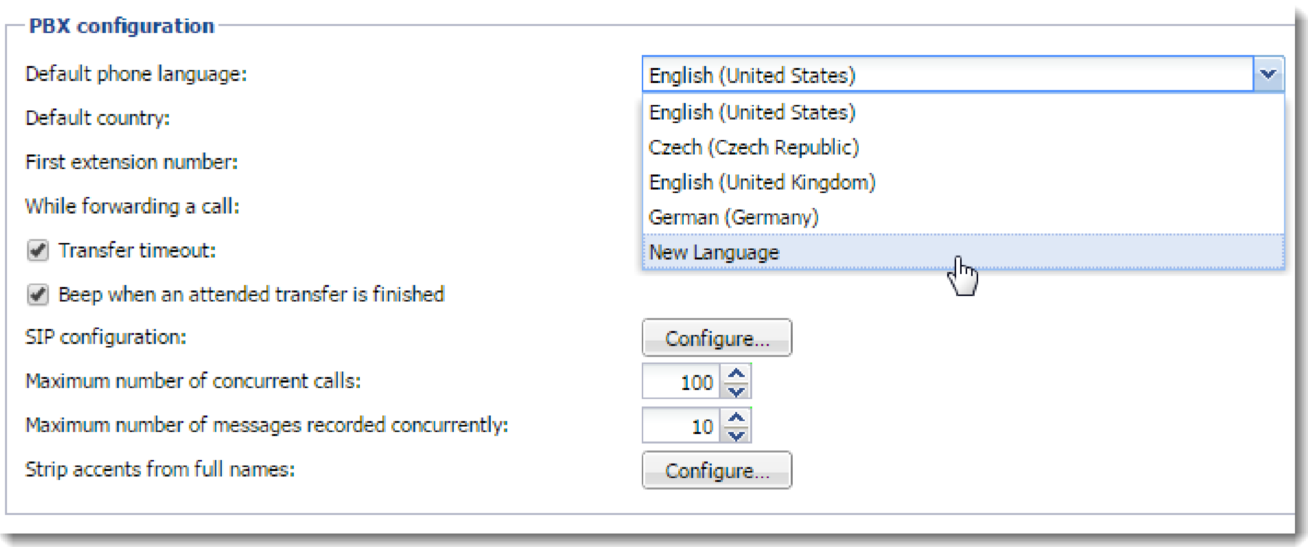
1. In the Kerio Operator administration interface, go to **Configuration > Advanced Options > Telephony**.
2. Next to the **Default phone language** field, click **Configure**.
3. In the **Voice Prompts** dialog box, click **Add**.
4. Type a name of the voice prompt.
5. Select a language syntax.
6. Click **Upload** and select your sound package.



7. Click **OK** and click **Close**.

8. In the **Default phone language** field, select the new language.

9. Click **Apply**.

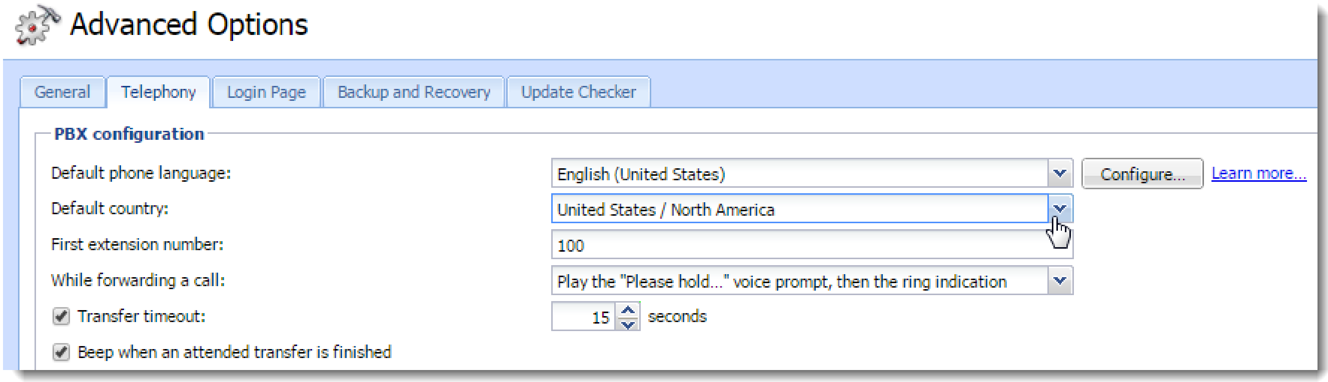


Changing indication tones according to countries

Each country has different standards for indication tones during calls (e.g. beeps, ringing tones, etc.).

You can change the settings in the administration interface.

To select a default country for your PBX, go to **Configuration > Advanced Options > Telephony**.



Example

Live And Let Laugh Inc has the following configuration:

» Joan Giggle, receptionist and operator, uses extension 100 and wishes the phone to communicate with her in the Happish language.

» Brian Snigger, receptionist and operator, uses extension 200 and is satisfied with the default language, which is English.

» Phoney VoIP, an interface for incoming calls, is configured in Kerio Operator with the default language — English. This interface is operated by Brian Snigger.

» Telephium VoIP, an interface for incoming calls, is configured in Kerio Operator for communication with customers from the Bliss Seekers Land (in Cravish). This interface is operated by Joan Giggle.

» The default language in Kerio Operator is English.

» Voicemail is enabled and the extension for accessing the voicemail is 50.

Scenario 1:

When Brian Snigger calls Joan Giggle (200 > 100) or when Brian Snigger calls the voicemail (200 > 50), the automatic announcements are in English.

Scenario 2:

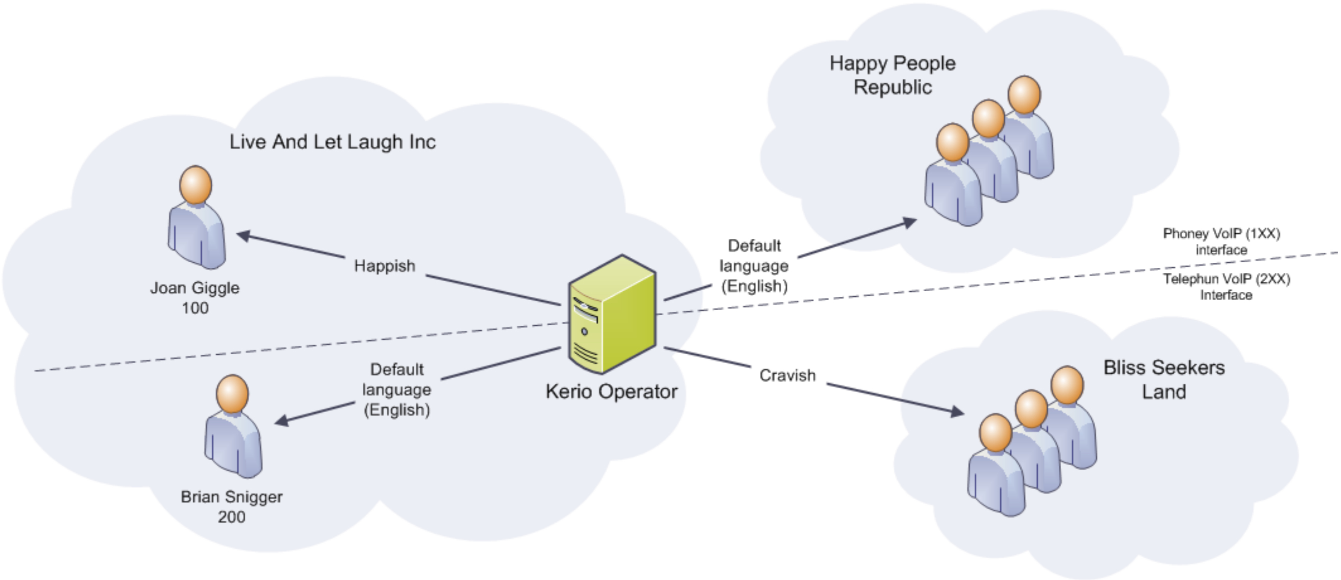
When Joan Giggle calls Brian Snigger (200 > 100) or when Joan Giggle calls the voicemail (200 > 50), the automatic announcements are in Happish.

Scenario 3:

Customers calling via the Phoney VOIP interfaces will hear announcements in the default language (English).

Scenario 4:

Customers calling via the Telephun VOIP interfaces will hear announcements in Cravish.



4.7.2 Configuring Built-in DHCP server in Kerio Operator

Kerio Operator includes a built-in DHCP server. There are deployment scenarios in which it is useful to have a separate DHCP server for VoIP devices:

» In larger networks, you may need a LAN segment dedicated to voice traffic.

» In smaller networks, the router/firewall sometimes does not support the DHCP option 66 for automatic provisioning of phones.

> **WARNING**
> DHCP server is disabled in the default mode so that it does not collide with your existing DHCP server.

Configuring DHCP server

The built-in DHCP server must have a static IP address:

1. In Kerio Operator administration interface, go to the **Network** section.

2. Select a network interface and click **Edit**.

3. In the **Interface Properties** dialog, switch configuration to **Use the following configuration** and type a new static IP address, mask and gateway.

4. Check **Enable DHCP server**.

5. Click **OK** to save the settings.

Kerio Operator will derive the configuration of the DHCP server from the values you set for the interface's IP address, network mask, and gateway. The DHCP server sends option 66 to Kerio Operator's own address with every address lease.

Assigning IP addresses

Kerio Operator generates the range of IP addresses from a configured mask of a network interface and assigns these addresses automatically.

Example:

» The configured IP address for a network interface is `192.168.62.1`

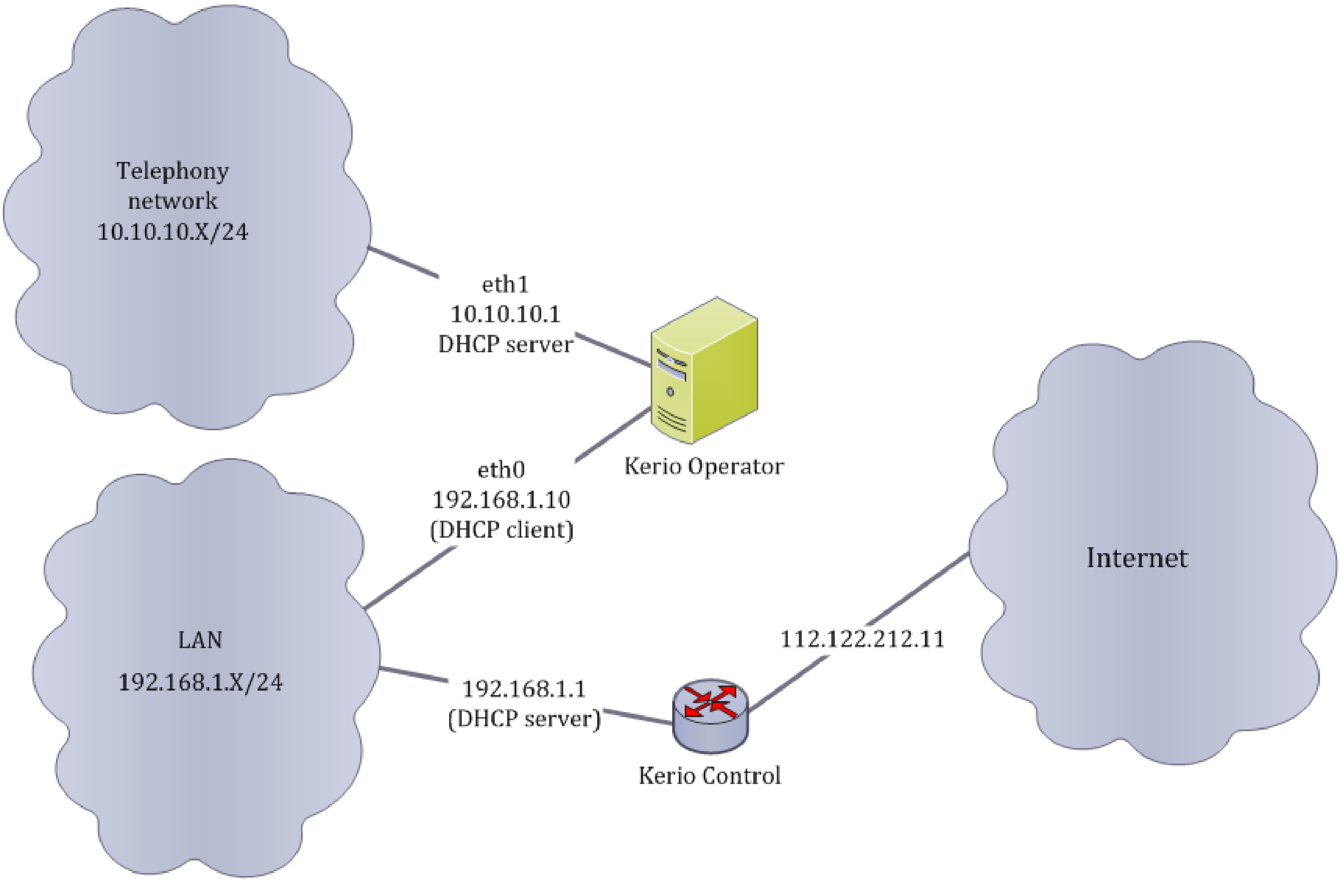
» The configured mask is `255.255.255.0`

» The gateway has the address `192.168.62.254`

In this example, the range of IP addresses is **192.168.62.2 — 192.168.62.253**.

Example — LAN segment is dedicated to voice traffic

In our example, you have LAN and you need to add an other network interface as a special telephony segment.

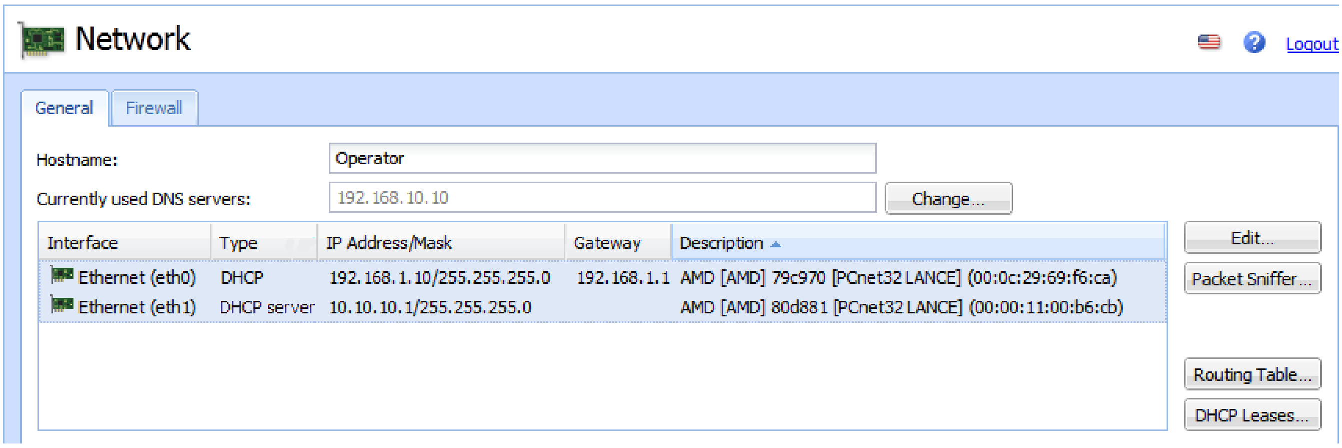


Screenshot 58: DHCP is running in the particular segment —scheme

You need to configure two interfaces in Kerio Operator administration interface:

1. Go to section **Configuration > Network > General**.

2. Configure interfaces as displayed bellow.



Screenshot 59: DHCP is running in the particular segment

4.7.3 Configuring parameter 66 in DHCP server in Kerio Control

The DHCP protocol assigns IP addresses. Apart from these addresses you can also send additional parameters via the DHCP protocol. Parameter 66 configures the TFTP server address.

How to set parameter 66 in Kerio Control

1. In the administration interface, go to section **DHCP server**.

2. If you use the automatically generated scopes, use **Click to configure scopes manually**.

3. Select a scope and open its settings (the **Edit Scope** dialog).

4. Click on the **Add** button.

5. Add parameter 66.

6. Enter a public IP address through which Kerio Operator communicates.

4.7.4 Configuring server date, time and time zone in Kerio Operator

Time Settings

Correct time and time zone settings of your PBX are necessary for correct configuration of telephone communication, time ranges and logs. If the time zone is not set properly, log messages or call history may contain confusing information. Therefore Kerio Operator is automatically synchronized with an NTP server.

> **WARNING**
> Do not change the settings unless you have a good reason.

NTP (Network Time Protocol) is a protocol for synchronizing time in your computer with time of the NTP server.

> **NOTE**
> Time and time zone settings on this tab refer to the administration interface time. It is the server time. Kerio Phone will display the time zone using the computer settings. If users are in a different zone to Kerio Operator, logs in call history will be displayed in users' time zone.

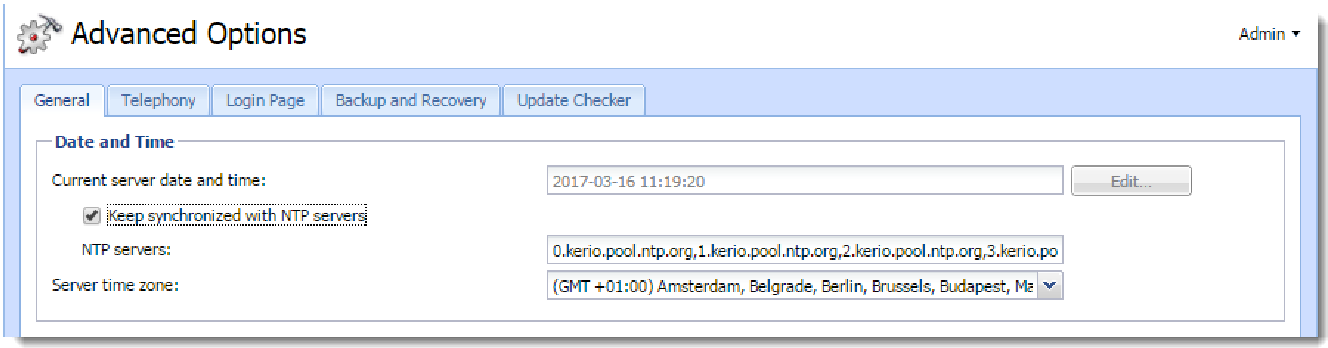
Configuring synchronization with NTP

1. In the administration interface, go to section **Advanced Options > General**.

2. Select the **Keep synchronized with NTP servers**. Date and time can be set manually but it is better to use an NTP server which provides information about the current time and allows automatic management of the firewall's system time.

3. Kerio Technologies offers the following free NTP servers for this purpose: 0.kerio.pool.ntp.org, 1.kerio.pool.ntp.org, 2.kerio.pool.ntp.org and 3.kerio.pool.ntp.org.

4. Click **Apply**.



Screenshot 60: Advanced Options — date and time settings

Configuring time zone

1. In the administration interface, go to section **Advanced Options > General**.

2. Select a time zone from the **Server time zone** list.

3. Click **Apply**.

The current date and time will be changed according to the new time zone.

4.7.5 Configuring standard phone interfaces

You can connect Kerio Operator to your provider using hardware cards.

You can use the card distributed with Kerio Operator Box series 3000 or you can use your own card and connect it to your Kerio Operator server.

Supported cards

Kerio Operator supports the following cards:

» PRI card — The number of concurrent calls varies depending on whether you have a contract with an American or European provider:

- T1 (in the USA) allows 23 concurrent calls.
- E1 (in the EU) allows 30 concurrent calls.

» BRI card — Has four ports, each of which can operate two concurrent calls.

» FXO card — Has four ports each of which can operate only one call at a time.

For a specific list of supported cards, see the Supported Phone Cards section on the Kerio website.

Prerequisites

Before you configure an interface, you need to know:

» Telephone number (or numbers) from your telephone provider

» (PRI/BRI only) Which ISDN type to use for communication. This usually differs by your location: for example, EuroISDN for the EU, Nation ISDN Type 2 for the USA, and so on)

» Whether your provider requires overlap dialing.

» Whether the provider sends or requires whole or abbreviated telephone numbers. For more information, refer to Mapping external and internal numbers (page 196).

» At least one internal extension defined in Kerio Operator (for example, the extension of an employee who redirects the calls).

Configuring interfaces

After connecting a card, configure the interface:

1. In the administration interface, go to the section **Configuration > Call Routing**. The **Interface and routing of incoming calls** table shows one of the following, depending on your card:

   - PRI card: one standard telephone interface
   - BRI or FXO card: four interfaces (one for each of the four ports)

2. Double-click an unconfigured interface. The configuration wizard opens.

3. Type a name for the interface (for example, your provider's name). The name must not contain spaces or special characters and must be unique.

One or multiple numbers

1. If you acquire one or multiple phone numbers from your provider, type the numbers in the **New provider > With external number** field. You can:

   - Separate individual numbers with commas (for example, 555450, 555451, 555452, and so on)
   - Type the whole range using – (for example, 555450-555459)

2. Click **Next**.

3. Select an extension to receive all calls from the provider.

4. (Optional) In the **Prefix to dial out** field, type a prefix for outgoing calls. Kerio Operator uses the prefix to route calls to your provider. This prefix can be same for other providers. For more information, refer to Working with prefixes for outgoing calls (page 202).

5. Click **Next**.

6. (PRI and BRI only) Select the **Switch type** in the dialog box:

   - If you are in the EU, select the EuroISDN option
   - If you are in the USA, select the National ISDN Type 2 option

7. Click **Next**.

8. Verify the information in the **Summary** section. If you need to add more information from your provider select the **Edit details of created interface** option. For more information, refer to Configuring additional details for an interface (page 272).

9. Click **Finish**.

10. Create a rewriting rule to correctly map numbers to internal user extensions.

Interval of numbers

1. If you acquire a trunk with an interval of numbers from your provider, type the numbers in the **New provider > With external number** field. Use `x` in place of the numbers that vary (for example, 555xxx).

2. Click **Next**.

3. Select an extension to which you want Kerio Operator redirect all calls to unassigned (unused) extensions.

4. (Optional) In the **Prefix to dial out** field, type a prefix for outgoing calls. Kerio Operator uses the prefix to route calls to your provider. This prefix can be same for other providers. For more information, refer to Working with prefixes for outgoing calls (page 202).

5. Click **Next**.

6. (PRI and BRI only) Select the **Switch type** in the dialog box:

- If you are in the EU, select the EuroISDN option
- If you are in the USA, select the National ISDN Type 2 option

7. Click **Next**.

8. Verify the information in the **Summary** section. If you need to add more information from your provider select the **Edit details of created interface** option. For more information, refer to Configuring additional details for an interface (page 272).

9. Click **Finish**.

10. Create a rewriting rule to correctly map numbers to internal user extensions.

Overlap dialing

Some telephone providers require telephone numbers as a whole, others require the telephone numbers one digit at a time. Ask your provider about their requirements. Follow these steps to configure the interface:

1. In the administration interface, go to the section **Configuration > Call Routing**.

2. Select an interface and click **Edit**. The **Edit External Interface** dialog opens.

3. Go to **Interface Card**.

4. Select the **Overlap dialing** option.

5. Click **OK**.

Configuring additional details for an interface

To set additional settings in your interface for incoming and outgoing calls:

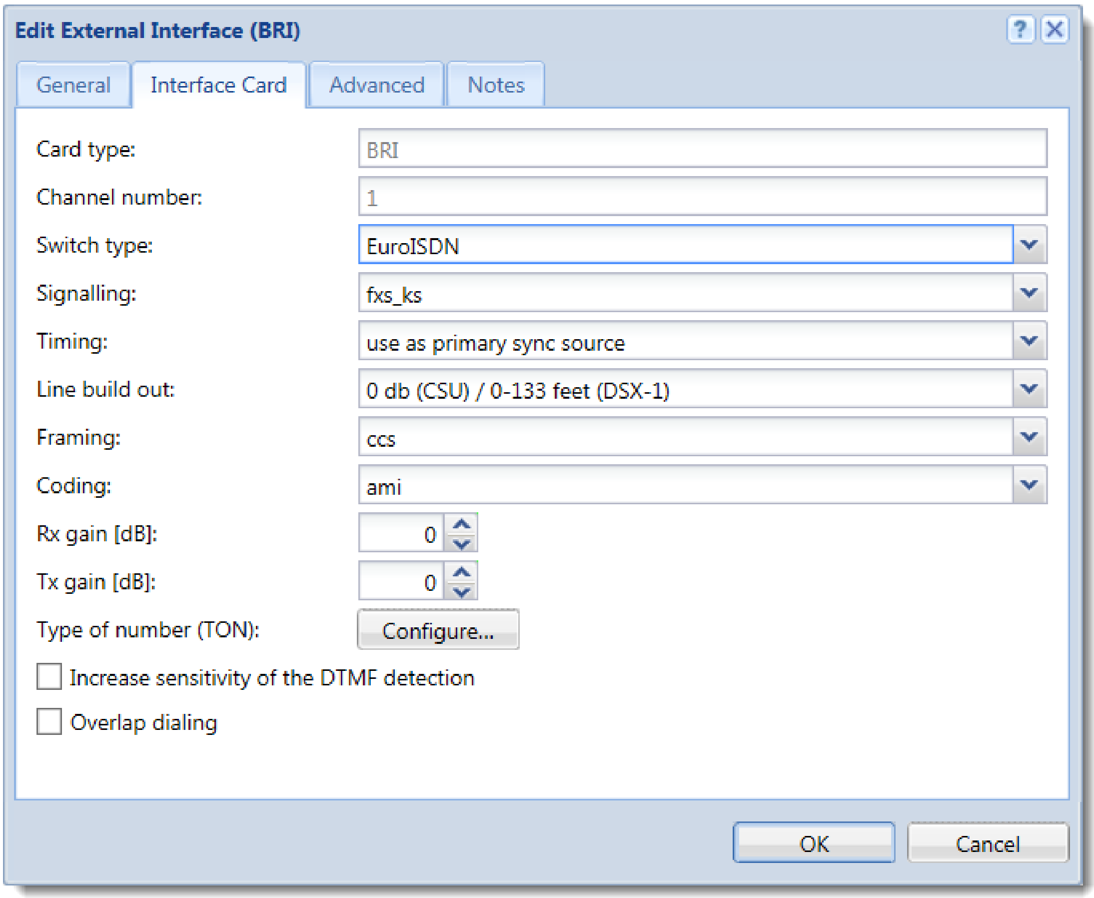
1. In the administration interface, go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.

2. Select an interface and click **Edit**.

3. On the **Interface Card** tab, change the interface settings. See the following chapters for details.

4. Click **OK**

> **NOTE**
> If you select the **Edit details of the created interface** option on the last page of the interface configuration wizard, this dialog box displays automatically.



Adjusting audio gain for standard phone interfaces

> **NOTE**
> New in Kerio Operator 2.4!

To adjust audio gain:

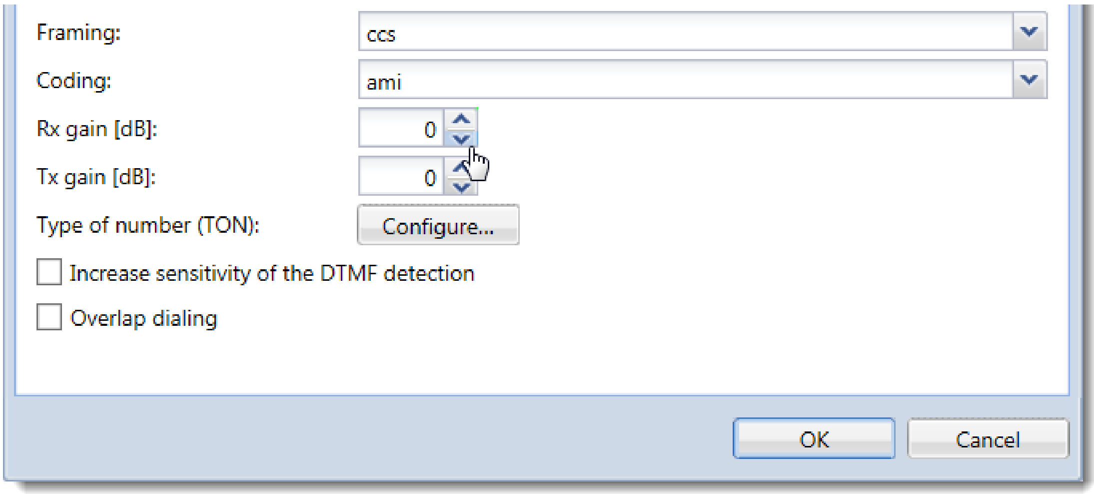
1. In **Configuration > Call Routing > Interfaces and routing of incoming calls**, select an interface and click **Edit**

2. Go to the **Interface Card** tab.

3. Set **Rx gain [db]**.

4. Set **Tx gain [db]**.

5. Click **OK**



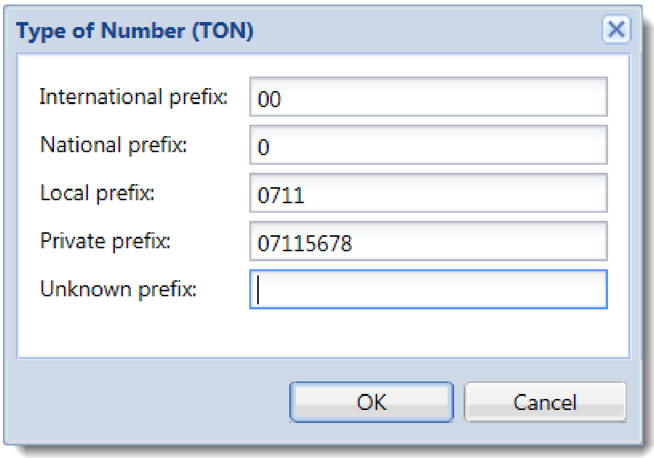
Configuring Type of number (TON)

> **NOTE**
> New in Kerio Operator 2.4!

Some providers send a stripped number with additional information about the type of the number. Kerio Operator can read these types and assign a prefix to the stripped number.

To configure prefixes for **Type of number (TON)**:

1. In **Configuration > Call Routing > Interfaces and routing of incoming calls**, select an interface and click **Edit**.
2. Go to the **Interface Card** tab.
3. Click **Configure** next to **Type of number (TON)**.
4. Type the prefixes you want to set.
5. Click **OK**



Increasing sensitivity of the DTMF detection

> **NOTE**
> New in Kerio Operator 2.4!

To enable this option:

1. In **Configuration > Call Routing > Interfaces and routing of incoming calls**, select an interface and click **Edit**.

2. Go to the **Interface Card** tab.

3. Select **Increase sensitivity of the DTMF detection**.

4. Click **OK**

Mapping of numbers

For more information, refer to Mapping external and internal numbers (page 196).

4.7.6 Connecting Kerio Operator to directory service

Which directory services are supported in Kerio Operator

Kerio Operator supports the following directory services:

- Microsoft Active Directory
- Apple Open Directory

What is the connection used for

In practice, mapping accounts from a directory service provides the following benefits:

Easy account administration

Apart from the internal database of user accounts, Kerio Operator can also import accounts and groups from an LDAP database. Using LDAP, user accounts can be managed from a single location. This reduces possible errors and simplifies administration.

Online cooperation of Kerio Operator and directory service

Additions, modifications or removals of user accounts/groups in the LDAP database are applied to Kerio Operator immediately.

Using domain name and password for login

Users may use the same credentials for Kerio Phone login and domain login.

> **WARNING**
> Mapping is one-way only, data are synchronized from directory service to Kerio Operator. Adding a new user in Kerio Operator createsa local account — it will not be duplicated into the directory service database.
>
> When creating user accounts in a directory service, ASCII must be used to specify usernames. If the username includes special characters or symbols, user may not be able to login to Kerio Phone or the administration interface.
>
> If you disable users in Microsoft Active Directory, they are also disabled in Kerio Operator (they will not be able to login to Kerio Phone, make or receive calls with their extensions).
>
> If you disable users in Apple Open Directory, they stay enabled in Kerio Operator.

Phone extensions can be managed in a directory service (if available) or locally in Kerio Operator. Select the most convenient option.

Connecting to a directory service

To map users from a directory service:

» Connect to directory service in section **Integration > Directory Service**.

» Activate users.

All information about directory services can be found in the **Config** log.

Microsoft Active Directory

In the administration interface, go to **Integration > Directory Service**.

1. Check the **Map user accounts from a directory service** option and select your directory service type.

2. In the **Domain name** field, enter the name of your Microsoft Active Directory domain — the domain name is then copied in other necessary fields.

3. In the **Hostname** field, enter the DNS name or IP address of the Microsoft Active Directory server. If you have a backup server, enter its name in the **Secondary hostname** filed.

4. In the **Username** and **Password** fields, enter the authentication data of a user with at least read rights for Microsoft Active Directory database. Username format is `user@domain`.

5. Within the communication of the Microsoft Active Directory database with the PBX, sensitive data may be transmitted (such as user passwords). For this reason, it is recommended to secure such traffic by using SSL. To enable LDAPS in Microsoft Active Directory, it is necessary to run a certification authority on the domain controller that is considered as trustworthy by Kerio Operator.

6. The rest of the items in the dialog are completed automatically. Do not change them unless you have a special reason to do so. These items are Microsoft Apple Open Directory domain name and Kerberos Realm which has to match the Microsoft Active Directory domain name, written in capital letters.

Apple Open Directory

In the administration interface, go to **Integration > Directory Service**.

1. Check the **Map user accounts from a directory service** option and select your directory service type.

2. In the **Domain name** field, enter the name of your Apple Open Directory domain — the domain name is then copied in other necessary fields.

3. In the **Hostname** field, enter the DNS name or IP address of the Apple Open Directory server. If you have a backup server, enter its name in the **Secondary hostname** filed.

4. In the **Username** and **Password** fields, enter the authentication data of a user with at least read rights for Apple Open Directory database. Username format is: `uid=root,cn=users,dc=domain,dc=tld`.

5. Within the communication of the Apple Open Directory database with the PBX, sensitive data may be transmitted (such as user passwords). For this reason, it is recommended to secure such traffic by using SSL. To enable LDAPS in Apple Open Directory, it is necessary to run a certification authority on the domain controller that is considered as trustworthy by Kerio Operator.

6. The rest of the items in the dialog are completed automatically. Do not change them unless you have a special reason to do so. These items are Apple Open Directory domain name and Kerberos Realm which has to match the Apple Open Directory domain name, written in capital letters.

Activating users from a directory service

Once the mapping is set, select individual users and map them to the PBX. This is how to map users:

1. Open the **Configuration > Users** section.

2. Click **Import > Import from a Directory Service**.

3. In the dialog, select all users you wish to map (you can also add users later) and click **Next**.

4. If users in the directory service have phone extensions assigned, you can either keep them or disable them. If you disable them, you have to assign new extensions. You can do it, for example, while changing your dial plan.

5. Click on **Finish**. Activated users are displayed in section **Configuration > Users**.

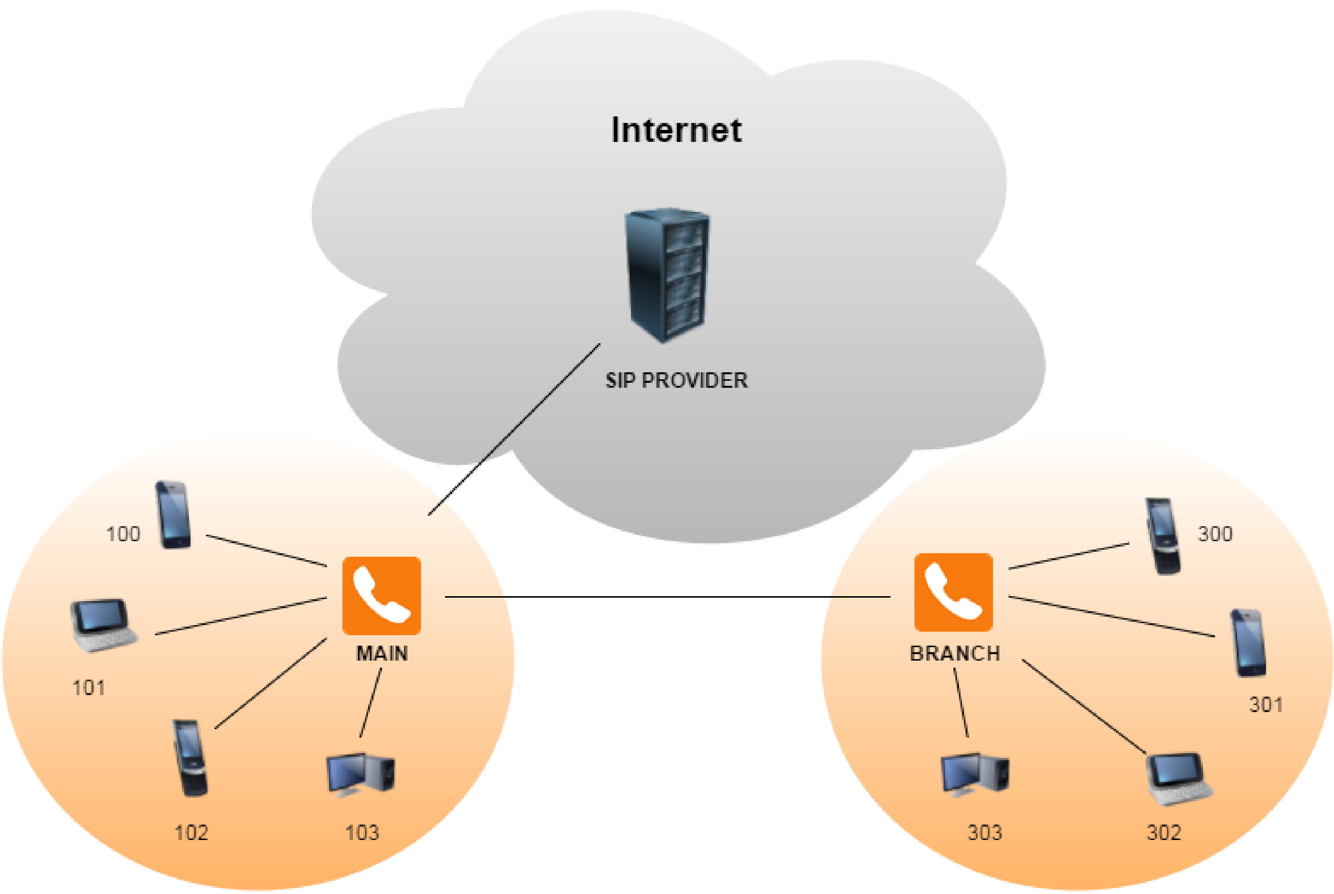
> **NOTE**
> Only extensions in attributes `telephoneNumber` (Microsoft Active Directory, Apple Open Directory) and `otherTelephone` (Microsoft Active Directory) can be mapped (are displayed). If you create special attributes in a directory service for your phone numbers, you will not be able to map such extensions.

4.7.7 Connecting multiple Kerio Operators

In Kerio Operator, you can connect multiple Kerio Operator servers. This enables you to directly reach remote phones by their extensions for free and send or receive external calls through a relay server.

The section below describes how to connect these two servers:

- The main server, which has internal extensions 100 — 199
- The branch server, which has internal extensions 300 — 399



> **NOTE**
> For more information about routing of calls between Kerio Operator servers and the PSTN, see Routing calls between multiple Kerio Operators and the PSTN.

Prerequisites

Before the start of the configuration, you need:

- Two Kerio Operator servers up and running
- Extension schemes for both phone networks, each of which has a unique set of extensions
- Both servers with public IP addresses or connected to the same network with a VPN tunnel

Connecting servers

On each Kerio Operator server, add a SIP interface for the other server.

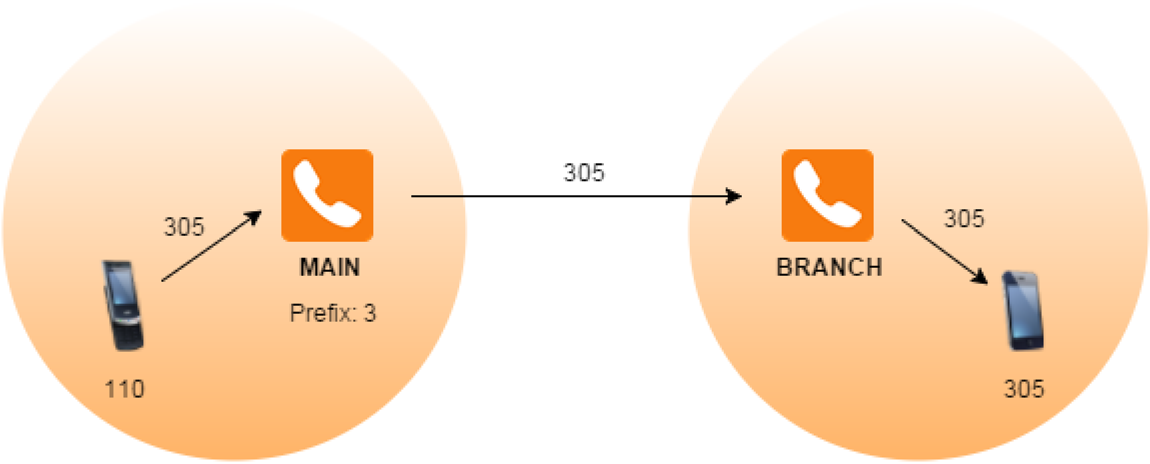
1. In the administration interface, go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.
2. Click **Add SIP interface**. The **Add SIP Interface** dialog box opens.
3. Type a name for the interface and select **Link to another PBX (without an external number)**.
4. Click **Next**.
5. In the **Prefix to reach the other PBX** field, type the appropriate number:
    - On the main server, type the prefix 3 (the first digit of each extension on the branch server)
    - On the branch server, type the prefix 1 (the first digit of each extension on the main server)
6. Click **Next**.
7. In the **Domain (IP address/hostname)** field, type the domain or the IP address:
    - On the main server, type the IP address of the branch server.
    - On the branch server, type the IP address of the main server.
8. Disable the **Required to register** option.
9. Click **Next**.
10. Verify the information in the **Summary** section.
11. Click **Finish**.

After the configuration of interfaces, Kerio Operator creates incoming and outgoing routes that use configured prefixes. These routes do not rewrite any numbers. Make test calls between the connected servers to reach their extensions.

Example of a test call

Call number 305 from extension 110 on the main server:

1. The user with an extension `110` dials number `305`.
2. Kerio Operator on the main server recognizes the prefix 3 and routes the call to the branch server.
3. The call arrives at the branch server and rings on the `305` extension.



4.7.8 Routing calls between multiple Kerio Operators and the PSTN

> **NOTE**
> Redesigned in Kerio Operator 2.4!

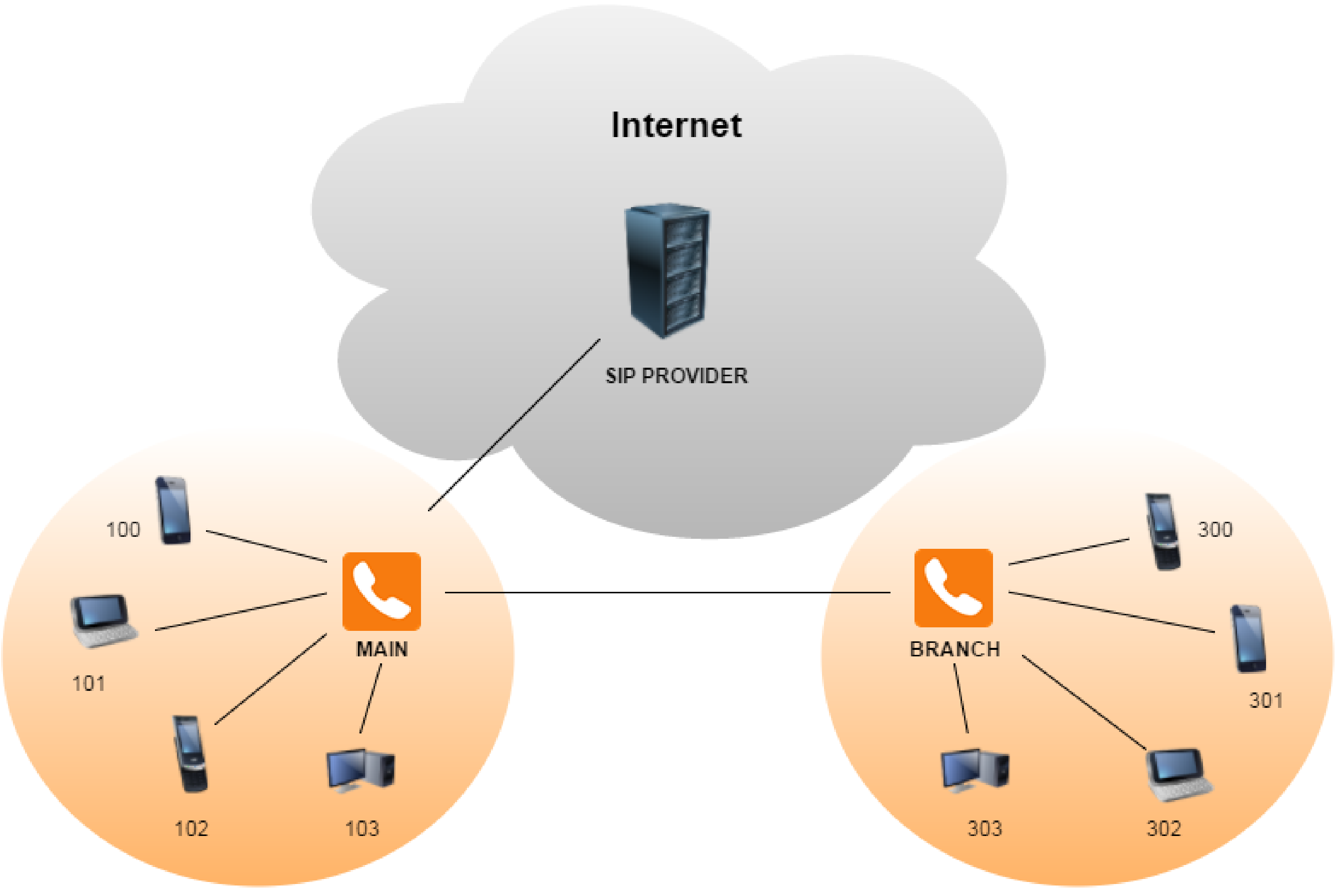
Learn how to:

- Reach the public switched telephone network (PSTN) from your connected Kerio Operator servers
- Route incoming calls from the PSTN to your branch servers

> **NOTE**
> For more information about connecting multiple Kerio Operator servers, see Connecting multiple Kerio Operators.

The sections below use the following example:

- Two connected Kerio Operator servers up and running:
  - The main server, which has internal extensions 100 — 199
  - The branch server, which has internal extensions 300 — 399
- Outgoing calls from the branch server to the PSTN go through the main server.
- Incoming calls from the PSTN to the branch server go through the main server.
- The prefix for outgoing calls to the PSTN is 0.
- External numbers from the SIP provider have the format 555 5xxx.



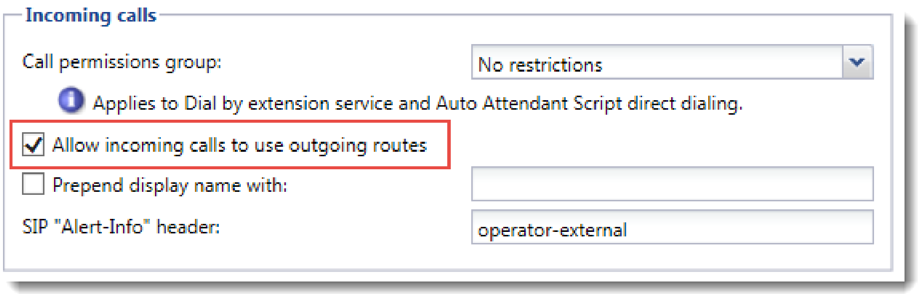
Calling to the PSTN through the main server

To call to the PSTN via the interface of the main server:

- Configure the interface on the main server.
- Create an outgoing route on the branch server.

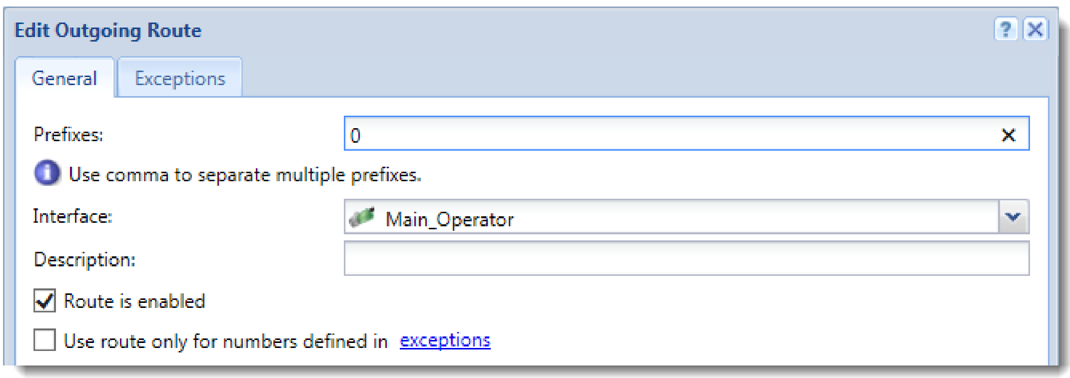
In the administration interface of the main server:

1. Go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.
2. Double-click the interface for the branch server.
3. Go to the **Advanced** tab.
4. Select the **Allow incoming calls to use outgoing routes** option.
5. Click **OK**



In the administration interface of the branch server:

1. Go to **Configuration > Call Routing > Routing of outgoing calls**.
2. Click **Add**.
3. Type the prefix for outgoing calls of the main server (0 in our example)
4. Select the interface of the main server.
5. Click **OK**

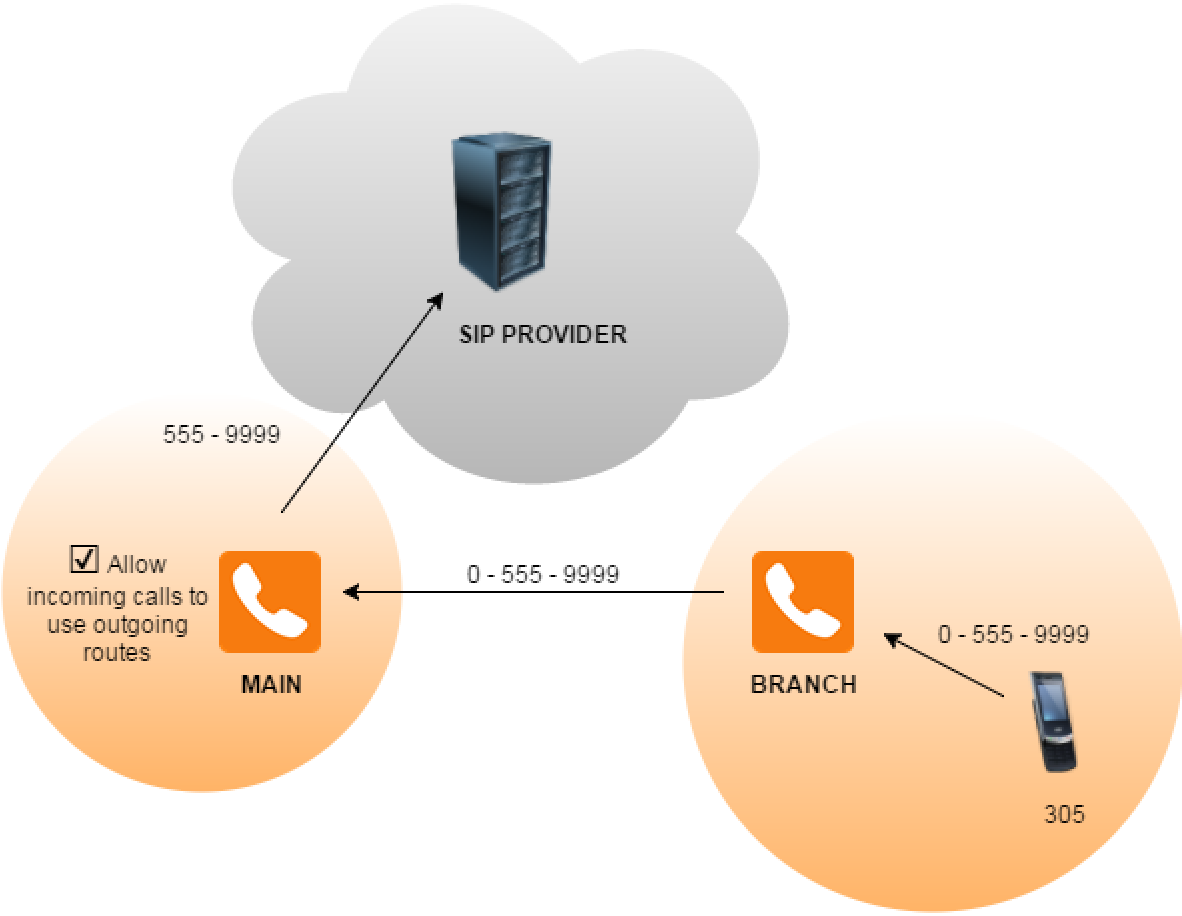


Make a test call to reach a number in the PSTN from the branch server.

Example of a test call

Call 555-9999 from extension 305:

1. The user with an extension 305 dials the number 0-555-9999.
2. Kerio Operator on the branch server recognizes the prefix 0 and routes the call to the main server.
3. The call arrives at the main server.
4. Kerio Operator on the main server recognizes the prefix 0 and strips the prefix off.
5. The main server routes the call to the SIP provider.



Routing incoming calls from the PSTN to the branch server

To route incoming calls to the branch server:

- If you have separate numbers, use speed dial extensions
- If you have a trunk of numbers, rewrite called numbers to match the internal extensions of the branch server

Using speed dial extensions

To use speed dial extensions:

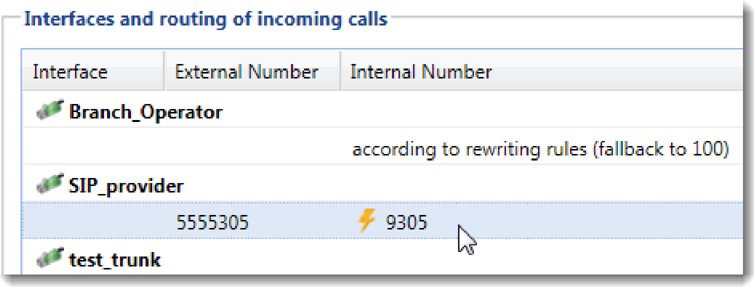
1. Create a speed dial extension (9305 in the example) that dials an extension of the branch server (305 in the example). For more details, see Creating and using speed dial.

2. Go to **Call Routing** and double-click the interface for the branch server.

3. Go to the **Advanced** tab.

4. (Optional) To enable users to return calls, select **Do not substitute the calling number when forwarding calls** and click **OK** This option also displays the caller ID of the caller instead of the number of the speed dial extension.

5. Double-click the number from your provider that you want to map.



Interfaces and routing of incoming calls

Interface	External Number	Internal Number
Branch_Operator		
		according to rewriting rules (fallback to 100)
SIP_provider		
	5555305	9305
test_trunk		

6. In the **Route incoming calls to** field, select the speed dial extension and click **OK**

7. Repeat steps 1—6 for all extensions you want to map to the branch server.

From now on, Kerio Operator uses the speed dial extension for all incoming calls that reach the external number and routes the call to extension `305` of the branch server.

Make a test call to reach an extension on the branch server.

Example of a test call

Call 555-5305 from 555-9999:

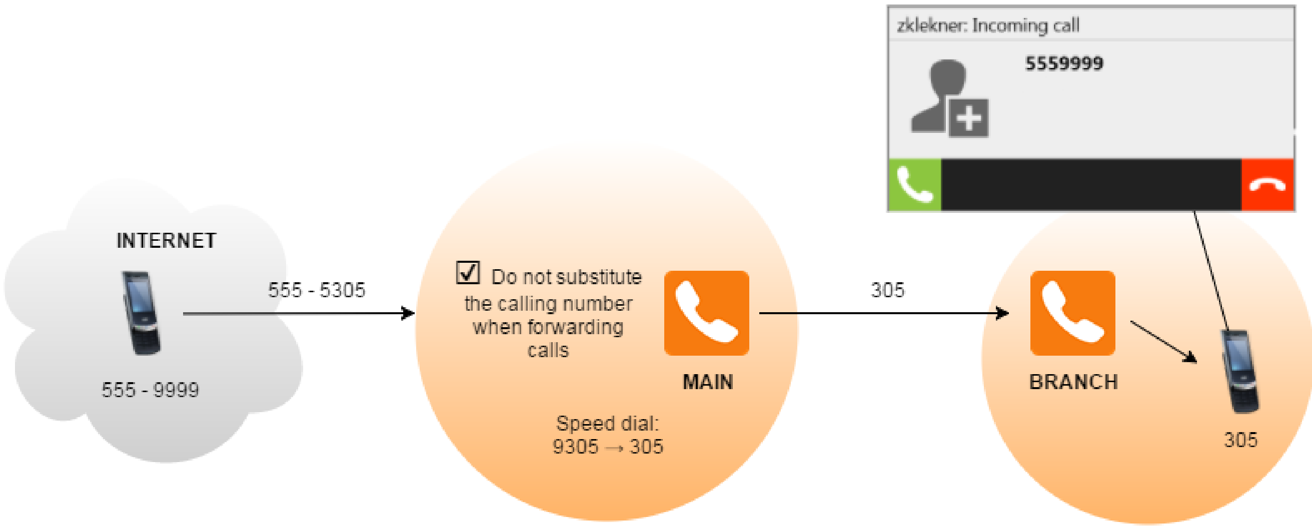
1. Caller dials the number `555-5305`.

2. The call arrives at the main server.

3. Kerio Operator routes the call to the `9305` extension and then to `305`.

4. The main server recognizes the prefix 3 and routes the call to the branch server.

5. The call arrives at the branch server and rings on the `305` extension.



Using number rewriting

To rewrite called numbers from your trunk and route them to the branch server:

1. In the administration interface of the main server, go to **Configuration > Call Routing > Interfaces and routing of incoming calls**.

2. Double-click the interface for the branch server.

3. Go to the **Advanced** tab.

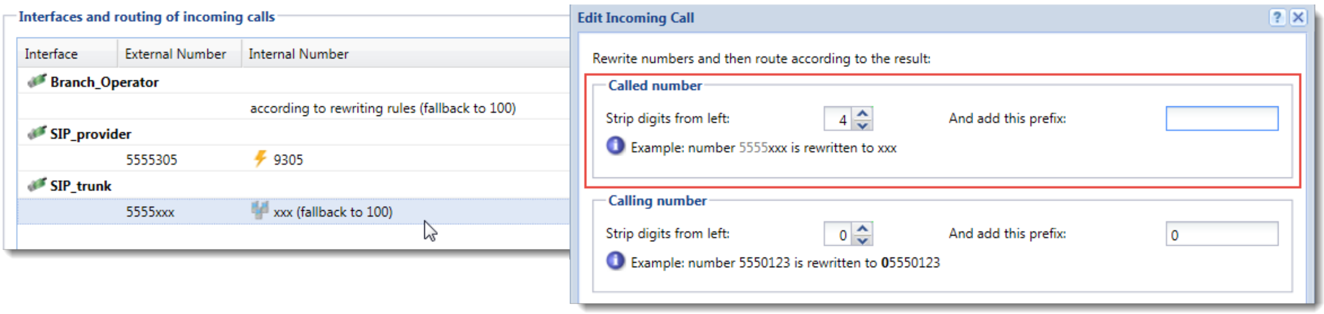
4. (Optional) To enable users to return calls, select **Do not substitute the calling number when forwarding calls** and click **OK** This option also displays the caller ID of the caller instead of the number of the speed dial extension.

5. Double-click the interface for your provider.

6. Go to the **Advanced** tab.

7. Enable the **Allow incoming calls to use outgoing routes** option.

8. Double-click the trunk of numbers to verify that Kerio Operator rewrites the called number correctly.



9. Click **OK**

After configuring the interface, make a test call to reach an extension on the branch server from the PSTN.

Example of a test call

Call 555-5310 from 555-9999:

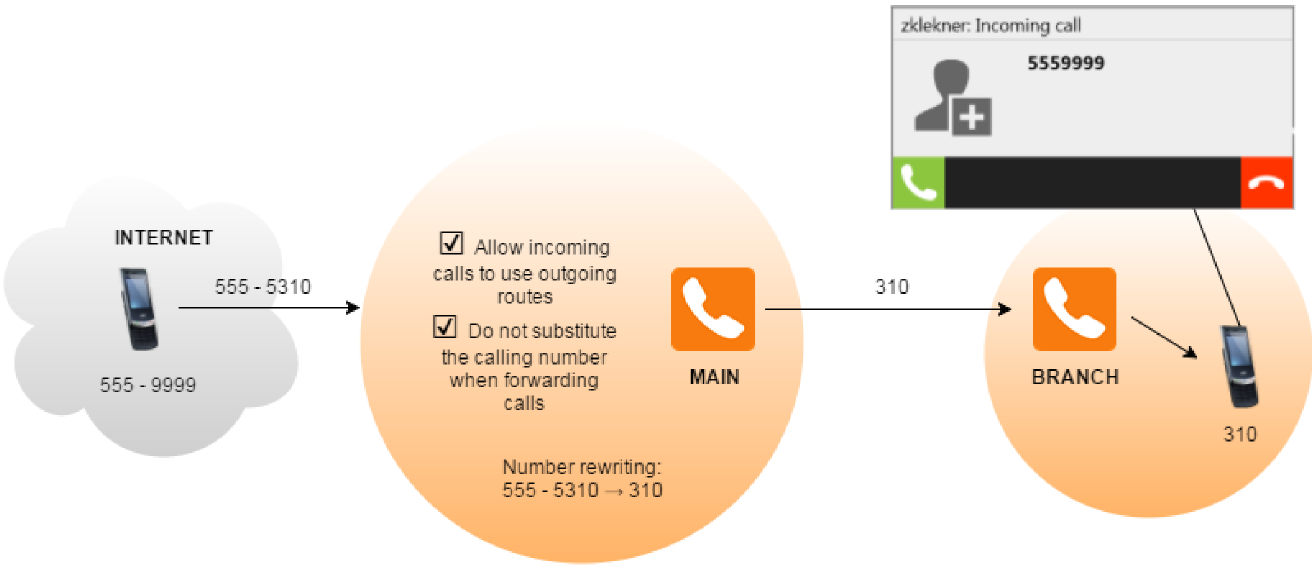
1. Caller dials the number `555-5310`.

2. The call arrives at the main server.

3. Kerio Operator matches the call to a SIP interface and strips off the first four digits of the number.

4. The call automatically uses the outgoing route with the prefix 3 and arrives to the branch server.

5. The call rings on the `310` extension.



4.7.9 Creating and using speed dial

Speed dial is a shortcut for phone numbers (for both the internal extensions and external phone numbers).

Adding speed dial

Before you begin creating speed dial, select a numerical range you will use. Speed dial must be different from current extensions. Generally, it is convenient to create speed dial so that they will not coincide with your dial plan in future.

1. Open **Speed Dial**.

2. Click **Add**.

3. In the **Add Speed Dial** dialog box, type a speed dial in the **Speed dial extension** field.

4. In **Dial number**, type the callee's phone number including the prefix for outbound calls.

5. Click **OK**

Configuring speed dial with DTMF

The speed dial with DTMF (Dual-tone multi-frequency signaling) is intended for calling special services like long distance phone service providers. If you need to place a call via such a service, you usually need:

- provider's number (usually it is a toll-free number that starts with `800`: 800555333)
- user ID (78901234)
- PIN (8808)
- a number you want to call (011420111222333)
- `#` character — denotes the end of the number and starts the call.

When you set the speed dial with DTMF, the number of steps is shortened to dialing the speed dial extension followed by the number you want to call: 89011420111222333

Configuring speed dial with DTMF

1. In the administration interface, go to **Speed Dial**.

2. Click **Add**.

3. In the **Add Speed Dial** dialog box, type a speed dial in the **Speed dial extension** field. In our example it is `89`.

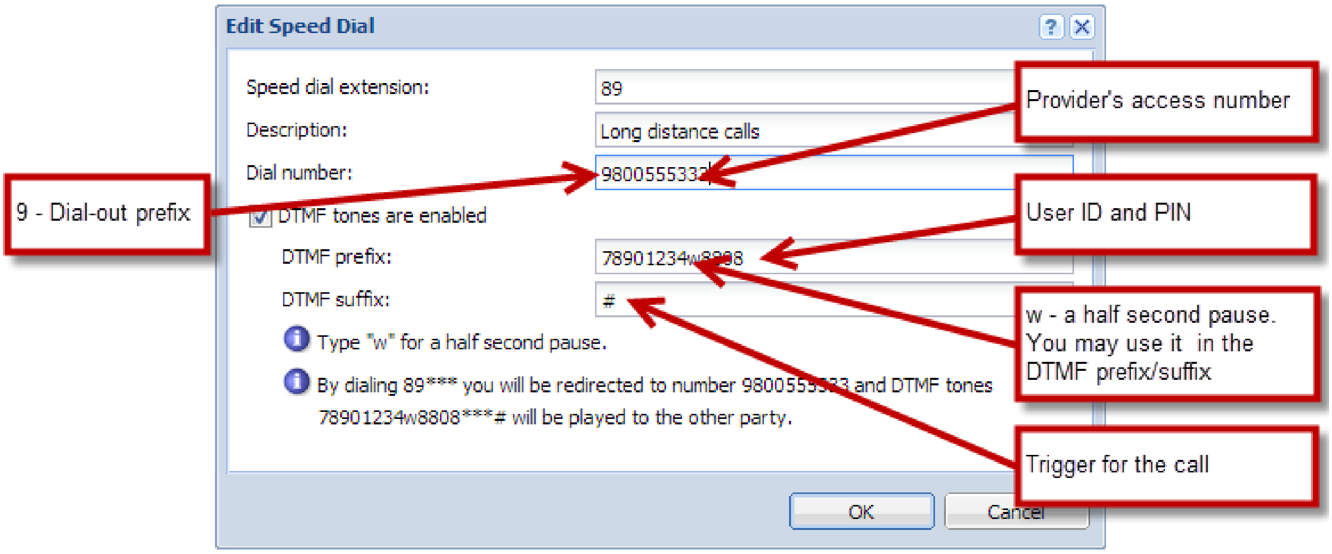
4. In **Dial number**, type the provider's access number including the prefix for outbound calls.

5. Select **DTMF tones are enabled**. Once you enable DTMF, the speed dial behaves as a dial-out prefix.

6. In the **DTMF prefix** field, type the access code and PIN. Your provider's IVR system may require a pause between typing the access code and PIN. Therefore use the `w` character for a half second pause. In our example it is `78901234w8808`.

7. In the **DTMF suffix** field, type `#`

8. Click **OK**



Screenshot 61: Add Speed Dial dialog

What is happen if you use the speed dial 89?

You want to call the number 011420111222333.

To place the call, you dial: 89011420111222333. The service will dial the access number 800555333 and once the call is connected, the following DTMF digits are sent:

789012348808 011420111222333 #

4.7.10 Creating ringing groups

You can use ringing groups to make calls ring simultaneously on multiple extensions.

Adding new ringing groups

1. In the Kerio Operator administration interface, go to **Ringing Groups** and click **Add**.

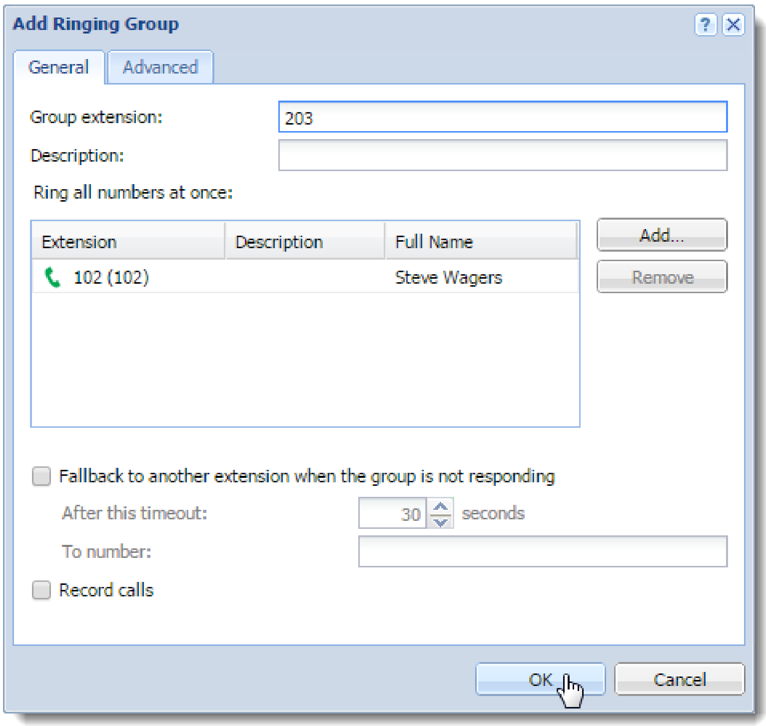
2. In the **Group extension** field, type the extension number for the group.

3. Add extensions you want to ring simultaneously to the table.

4. (Optional) To redirect the call to another extension when no one answers the phone, select **Fall back to another extension when the group is not responding** and set a timeout and destination extension.

5. (Optional) If you don't want to display the answered call as missed on other phones in the group, go to **Advanced** tab and select **Do not display missed calls on the phones**.

6. Click **OK**



4.7.11 Customization of voice sets

This summary provides information on how to customize/change voice sets in Kerio Operator.

The Internet provides many sources of localized and customized basic sounds and voice prompts. Voice sets for various languages can be found at http://www.voip-info.org. However, it is recommended to use voice sets present in Kerio Operator than downloading them from the Internet, as it may not include all of the prompts used by Kerio Operator.

To customize voice sets:

1. Log in to Kerio Operator administration. For more information, refer to Logging into Kerio Operator Administration (page 19).

2. Go to the **System Health** dialog, hold the **shift** key and click **Tasks**. You should see an option to enable **SSH**.

3. Log in to your system with an SSH client (using root and your admin password).

4. Go to **/var/lib/asterisk/sounds**.

5. Download the directory of sounds you will use as a base (e.g. en_GB).

6. Modify the .gsm sound files using any compatible sound editor.

7. Compress the folder.

8. Back in the Admin Console, open **Advanced Options > Telephony**.

9. Click **Configure** next to **Default phone language** option and upload the voice sets.

Once you upload a voice and sound set, you can use it for Kerio Operator, individual interfaces or individual users. For more information, refer to Language settings in Kerio Operator (page 262).

4.7.12 Customizing the Kerio Phone login page

Adding your custom logo

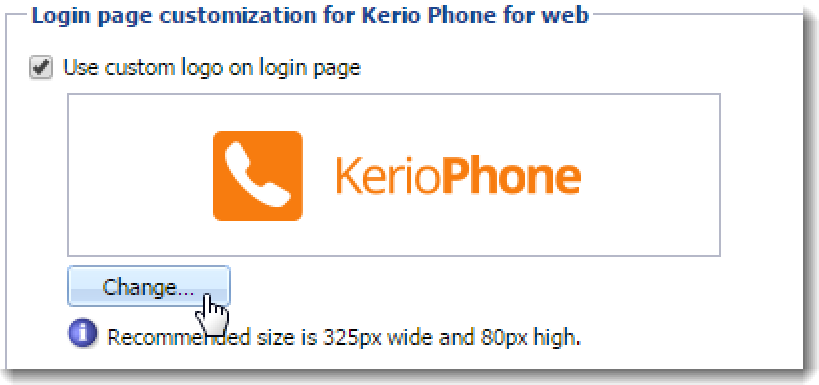
To change a logo of your login page:

1. In the administration interface, go to **Configuration > Advanced Options > Login Page**.

2. Select the **Use custom logo on login page** option.

3. Click **Change** and locate the new logo file. The logo must be in the PNG format. The recommended maximum size is 325 x 80 pixels.

4. Click **Apply** to save your settings.



Configuring your custom button style

To change a style of a button:

1. In the administration interface, go to **Configuration > Advanced Options > Login Page**.

2. Select the **Use custom button style** option.

3. Type a color's hex value for **Text color** (for example, `#ffffff`).

4. Type a color's hex value for **Background color** (for example, `#669900`).

5. Click **Apply** to save your settings.

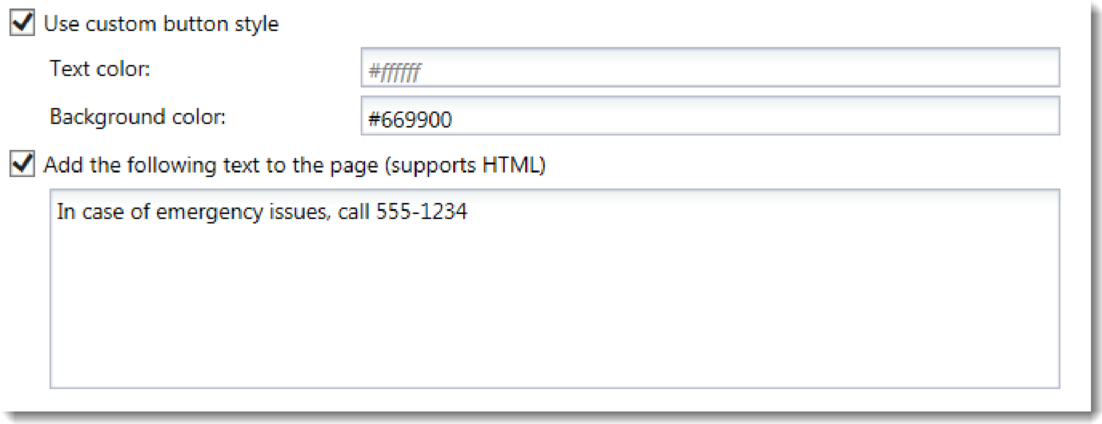
Adding your custom text

To add a text to your login page:

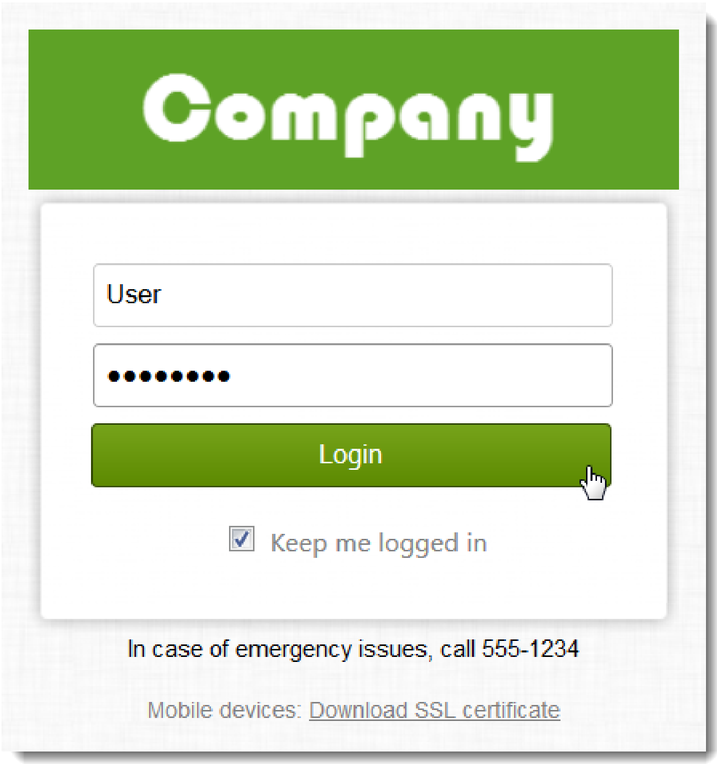
1. In the administration interface, go to **Configuration > Advanced Options > Login Page**.

2. Select the **Add the following text to the page (supports HTML)** option.

3. Type your text (for example, **In case of emergency issues, call 555-1234**).



4. Click **Apply** to save your settings.



4.7.13 Distinctive ringing support

Kerio Operator supports setting different ring tones for different types of calls (external calls, internal calls or ringing groups).

Configuring strings Kerio Operator

> **NOTE**
> New in Kerio Operator 2.5!

By default, Kerio Operator uses the following strings for the Alert-Info header:

» `operator-external` (calls from an interface)

» `operator-queue` (calls from a call queue)

» `operator-group` (calls to a ringing group)

To configure different ring tones for your SIP interfaces, call queues and ringing groups, change the default string in **SIP "Alert-Info" header**.

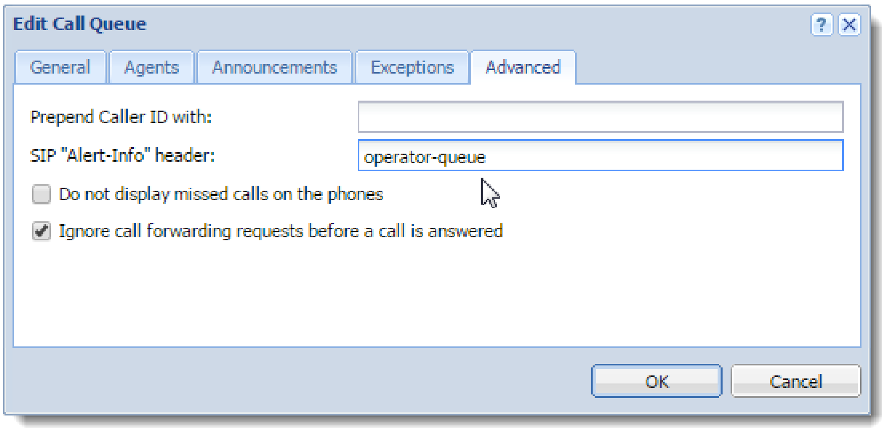
1. Go to **Call Routing**, or **Call Queues**, or **Ringing Groups**.

2. In **Call Routing**, double-click a SIP interface. In **Call Queues** or **Ringing Groups**, double-click an extension.

3. Switch to the **Advanced** tab.

4. In **SIP "Alert-Info" header**, change the default string.

5. Click **OK**



Configuring telephones (example: snom 360)

1. Go to web administration of your telephone.

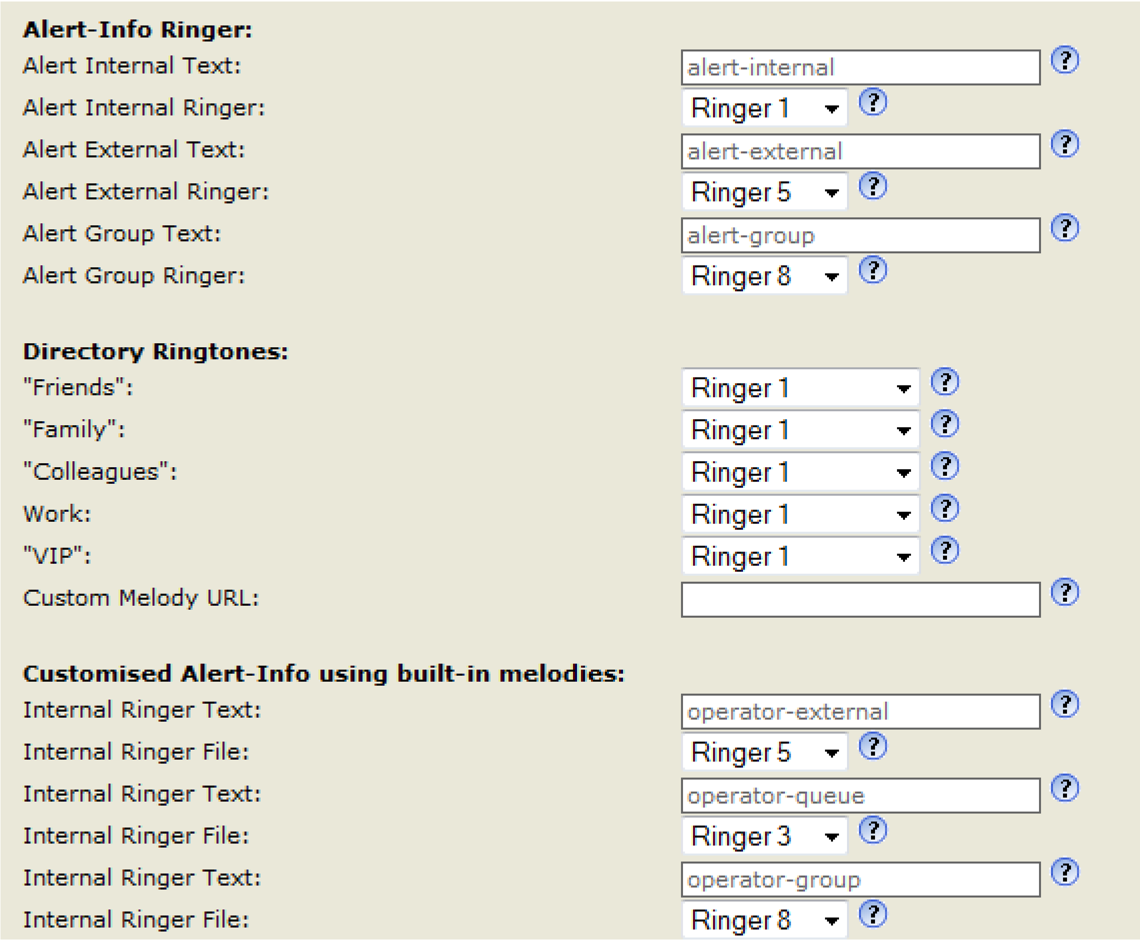
2. Go to **Setup > Preferences**

3. Find the alert-info settings.

4. Set different ringers for different alert-info strings (see screenshot).

5. Save the settings.

For testing purposes: Try to make a call from an external telephone number, from an internal extension and to ringing group.



Screenshot 62: Customising Alert-Info strings

4.7.14 Fax support in Kerio Operator

Using fax in Kerio Operator

Kerio Operator supports:

» T.38 protocol

» Fax-to-email

» PDF-to-fax

T.38 support

T.38 is a protocol for realtime transmission of fax over IP.

Kerio Operator uses T.38 by default. Ask your provider whether they support this protocol. If not, read section My provider does not support T.38.

Connecting a fax machine to Kerio Operator

1. Connect your fax machine to an Analog Telephone Adapter device (ATA — for example, Cisco SPA 112).

2. Assign one of Kerio Operator extensions to the ATA device.

Fax machine is connected to the network. You can send and receive faxes.

Configuring an ATA device

You can use various ATA devices. Each device has different settings. The following must be configured:

1. enable T.38

2. set **fax passthru** to **ReInvite**

> **NOTE**
> Phone provisioning in Kerio Operator sets these variables automatically.

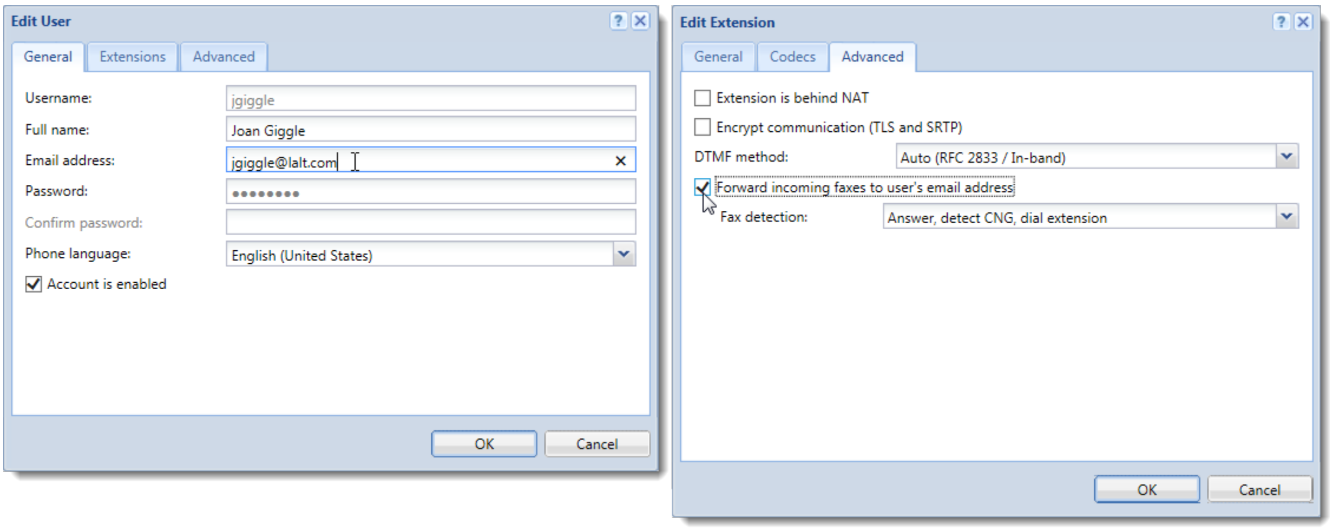
Receiving faxes to a user's email address

You can enable fax-to-email service for any extension. Kerio Operator then sends all incoming faxes to the user's email address as PDF attachments.

> **WARNING**
> In the administration interface, define SMTP relay in section **Advanced Options > General** so that your Kerio Operator can send emails.

In the administration interface:

- go to **Users** and enter an email address for each user.
- go to **Extensions** and enable option **Forward incoming faxes to user's address** for the particular user's extension.



Configuring fax detection (CNG signal)

A CNG signal is the fax machine sound you may hear when there is a fax machine connected to the other end of line. Kerio Operator can detect the signal and start receiving faxes automatically.

1. In the administration interface, go to **Extensions**.

2. Double-click a selected extension.

3. On tab **Advanced**, select:

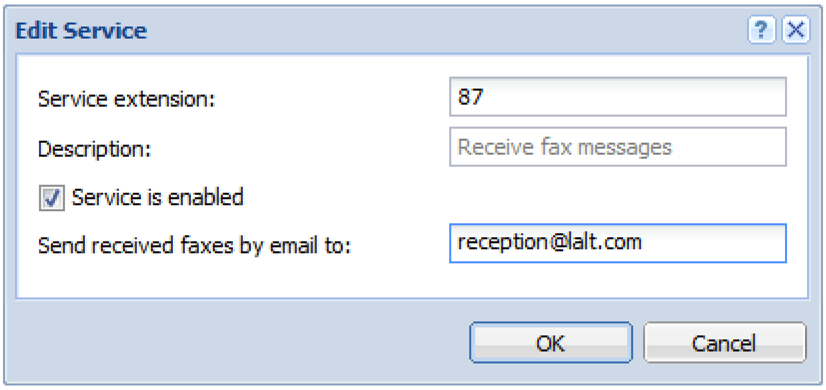
- **Dial extension, wait for answer, detect CNG** — PBX dials an extension, waits for an answer and then starts detecting the CNG signal. User has to answer a call first in order to receive faxes. When a fax tone is detected, the call will be taken over by Kerio Operator.
- **Answer, detect CNG, dial extension** — PBX answers a call first, then detects the CNG signal and immediately dials an extension. If users don't answer the phone, a fax mail is received and users have a missed call on their phone display. This option is good for occasional fax transmissions.
- **Answer, detect CNG, wait 3.5 seconds, dial extension** — Extension is dialed after a 3.5 seconds delay which is used to detect faxes. There will not be any missed calls shown on the phone's display. Regular calls will be automatically answered and will be followed by a 3.5 second delay of silence. This option is good for more frequent fax usage.
- **Answer, detect CNG, wait 3.5 seconds (ringing tone), dial extension** — the PBX will generate a ringing tone instead of waiting in silence. This option is also good for more frequent fax usage and may be less confusing to human callers.

4. Save the settings.

Receiving all faxes to a specific email address

Kerio Operator can send all incoming faxes to a single email address.

1. Go to **PBX Services**.
2. Open **Receive fax messages**.
3. Type email address in the **Send received faxes by email to** field.



Kerio Operator will send all incoming fax messages to the specified email address.

My provider does not support T.38

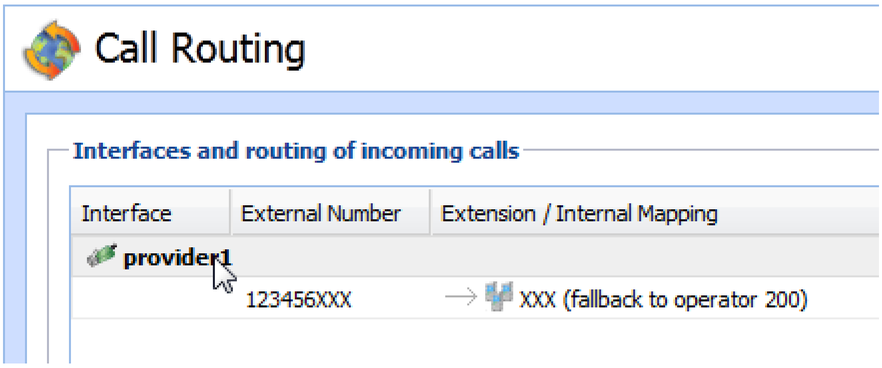
WARNING
Fax support without T.38 is not reliable. Using codecs G.711 A-law/U-law instead of T.38 is a workaround.

If your SIP provider does not support T.38, you have to solve these issues:

- Enable codecs G.711 A-law/U-law for the transmission. High compression codecs would distort signal.
- Reduce the speed on your fax machines (if supported).

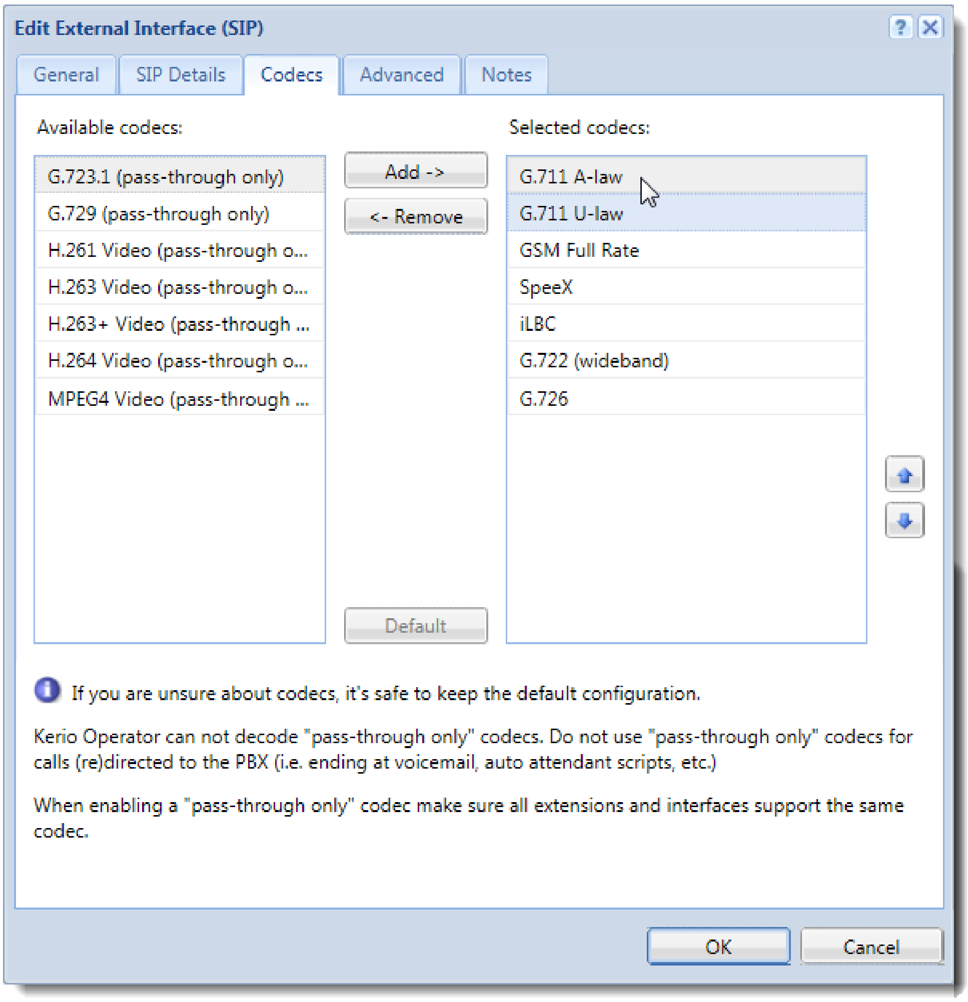
Enabling G.711 A-law/U-law codecs for the interface

1. Login to the administration interface.
2. Go to **Configuration > Call Routing**.
3. Click the provider's interface.



4. Click the **Codecs** tab.
5. Move **G.711 A-law** and **G.711 U-law** to the **Selected codecs** table.
6. Move **G.711 A-law** and **G.711 U-law** codecs up in the table.

> **NOTE**
> Moving G.711 A-law/U-law codecs up in the table can cause bandwidth consumption.



7. Click **OK**

Fax messaging now uses codecs G.711 A-law/U-law.

Disabling the T.38 support

Although your SIP provider supports T.38 protocol, you may experience some difficulties in communication. Conclusion is disabling a support of the T.38 protocol:

1. In the administration interface, go to **Configuration > Advanced Options**.

2. On the **General** tab, click **Configure...** next to the **SIP Configuration**.

3. Unselect **Use T.38 standard for faxing**.

For more information, refer to My provider does not support T.38 (page 293).

Sending PDF to fax

For more information refer to Sending PDF to fax in Kerio Phone.

4.7.15 Hosting Kerio Operator

In some situations, it may be preferred to deploy Kerio Operator at a remote site, or data center, as these locations may offer better bandwidth, reliability, and consolidated management. This topic addresses the considerations when deploying Kerio Operator outside of the local network.

Phone Provisioning

When Kerio Operator is deployed on the same network as the IP phones, the provisioning process can be handled automatically through DHCP. However, if Kerio Operator is remote, the phones typically must be configured manually. Consider pre-configuring the phones at a convenient location before deployment at the remote site. For ongoing maintenance of the phones, it will be necessary to use the web administration of the phones, which will require access to the remote network. We've found Snom, Yealink, and Linksys phones to offer the best options for web configuration and remote management. For added security, automatic provisioning should be disabled from the dialog in the web administration, located under Provisioned Phones > Phone Provisioning.

With the introduction of the Kerio Phone mobile app, configuration from mobile devices is quite easy, and ideally suited in case Kerio Operator is in a hosted, or remote environment. For more information, refer to Provisioning of Kerio Operator Softphone for mobile devices (page 175).

Network Configuration

Because Kerio Operator includes a built-in firewall, it's not necessary to incorporate an external firewall. If possible, Kerio Operator should be assigned an Internet routable IP address to avoid network address translation, as this can cause a variety of call quality, or connectivity type issues. In many cases, the phones will be connecting through NAT, and there is an option in the extension properties to account for this. For more information, refer to Configuring NAT (page 259).

Bandwidth

When Kerio Operator is remote to the phone network, it means that local calls between extensions will be routed through the Internet. It's important to take this into consideration when evaluating the bandwidth requirements for a remote network of phones. A typical phone call may consume about 80 kbps, although some codecs may consume less bandwidth. For locations with limited bandwidth, consider setting GSM as the preferred Codec for those extensions provisioned on that network. For more information, refer to Bandwidth used by the different codecs (page 208).

Security

In a hosted scenario, Kerio Operator is typically accessible directly over the Internet (unless access is restricted through VPN). It's therefore necessary to pay close attention to the security settings related to Kerio Operator. For more information, refer to Securing Kerio Operator (page 254).

4.7.16 Setting optional call recording

Call Recording

Call recording is subject to special laws in many countries and may not be legal in your jurisdiction, or may require notice to the other party of the call.

In Germany and maybe in other countries you have to give callers (your customers) an option to continue without having their call recorded. This topic helps you with the following simple settings:

- Creating two call queues, one with call recording, the second without recording.
- Preparing voice prompt (something like "This call is recorded ..."). How to easily create voice files you can find out in Using PBX services topic.
- Creating simple auto attendant script which allows your customers to choose call queue without call recording.
- Recorded calls backup.

How to create and configure call queues

1. In the administration interface, go to section **Configuration > Call Queues**.

2. Open the **Add Call Queue** dialog and enter the extension number for the new queue on tab **General** (for example 400).

3. Description could be Calls are recorded.

4. Click on **Record calls**.

5. Set other parameters according to the Configuring call queues topic.

6. Open the **Add Call Queue** dialog again and enter another extension number on tab **General** (for example 401).

7. Description could be Calls are not recorded.

8. Set remaining parameters just as the first call queue.

9. Do not click on **Record calls**.

> **NOTE**
> Agents need to logged to both queues.

Now, you have two equal call queues. One of them is recorded and the other is not recorded.

How to create auto attendant script for two call queues

1. In the administration interface, go to **Configuration > Auto Attendant Scripts**.

2. Click **Add**.

3. Enter script extension (for example 500).

4. In the **Edit Menu** dialog, click **Edit**.

5. Add some description (script for recorded and unrecorded call queues).

6. Select prepared announcement (for example: "This phone call will be recorded. If you do not want recording your call, please, press 1 and wait for the connection.")

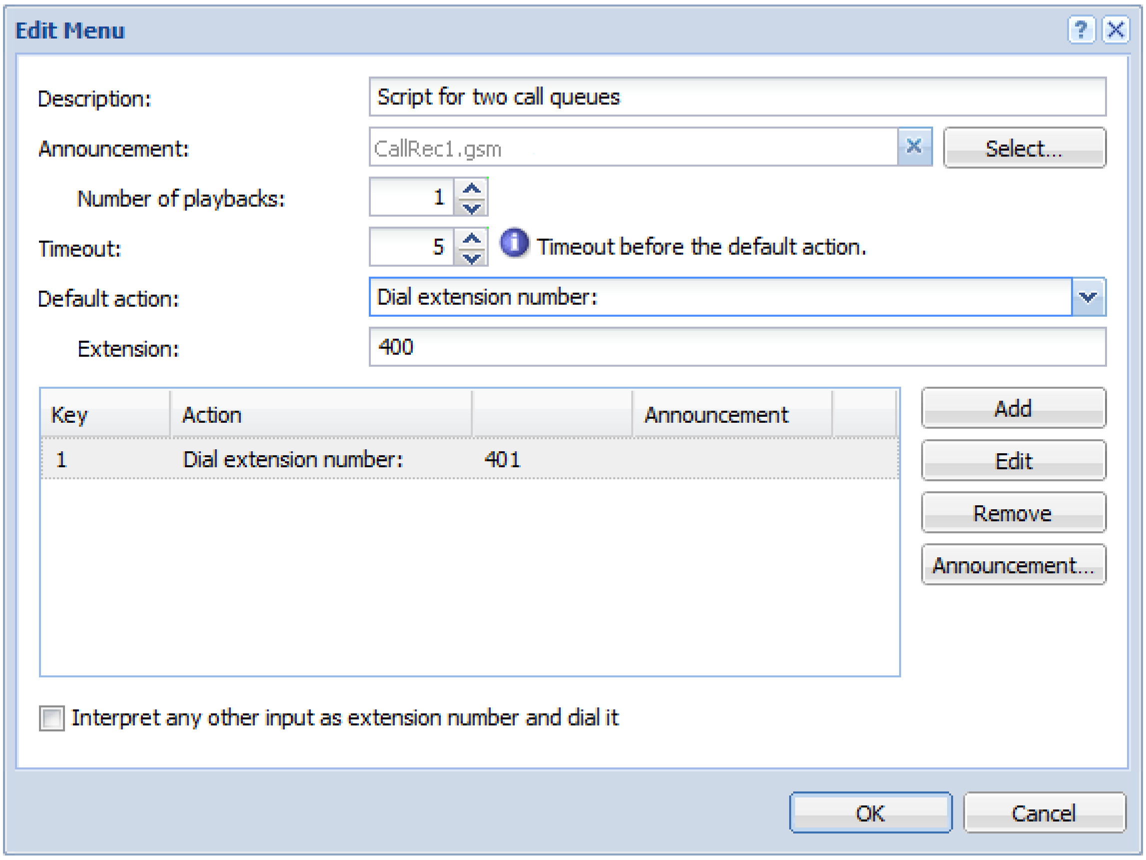
7. Set **Default action** to **Dial extension number** and enter extension of the call queue with call recording.

8. Create a new row by clicking on **Add**.

9. Key is set to 1.

10. Set **Action** to **Dial extension number** and specify the extension of the call queue without call recording.

11. Click **OK**



For more information, refer to Configuring auto attendant scripts (page 230).

Saving recorded calls

Kerio Operator can automatically backup recorded calls:

- locally — default settings
- at a remote storage

Saving recorded calls at a remote FTP/SFTP storage

1. Configure a FTP or SFTP server for storing the backup. For more information, refer to Configuring Remote FTP/SFTP Storage (page 304).

2. In administration interface, go to **Status > Recorded Calls**, click **Settings**.

3. Select **Save to remote storage**.

4. Select a type to **FTP/SFTP**.

5. Verify the **FTP/SFTP** URL.

6. Click **Test Connection**.

4.7.17 Setting outgoing calls constraints in Kerio Operator

You may want to limit some or all outgoing calls for a variety of reasons. For example, should an outside party obtain the username and password of one of your employees, they could use your PBX for international calls—possibly involving fraud and costing you money. It is therefore critical to have calls to external networks well configured.

You can set outgoing call constraints to prevent these types of attacks.

Restricting the length of individual outgoing calls

To set the maximum call duration:

1. In the administration interface, go to **Configuration > Security**.

2. Set **Maximum duration of each outgoing call**. The recommended value is 2 hours.

Restricting the number and length of outgoing calls

You can limit all outgoing calls by creating special rules in the section **Configuration > Security** in table **Outgoing calls constraints**.

The default rule limits the number of outgoing calls to 50 per hour and total call duration to 2 hours per day.

Example

A manufacturer in the United States sells and primarily has contacts just in the U.S. and Canada, but has a factory in Mexico. Management wants to limit calls to other countries.

1. In the administration interface, open **Configuration > Security** and click **Add**.

2. Type a rule name, such as `Constraints for Mexico`).

3. In the **Apply to these outgoing calls** section, select **All except listed** and click **Add**.

4. Add the calling prefixes as a single string:

   - For local calls: 9 (outside line)
   - For U.S. and Canada: 91 (outside line + 1 preceding the area code)
   - For Mexico: 901152 (outside line + 011 for international call + 52 for Mexico's country code)

5. Define the conditions: Set **Maximum calls count** to `10` per hour and **Maximum total calls duration** to 1 hour a day.

6. When the conditions are met, Kerio Operator can send a warning email or block all outgoing calls.

We recommend creating:

» One soft rule with lower limits that sends warning messages via email.

» Another rule with higher limits that blocks the PBX.

> **WARNING**
> If the limits are reached and the PBX is blocked, no one will be able to make calls to the restricted prefixes. However, an administrator can unlock the PBX in section **Configuration > Security**. We recommend making a thorough analysis of your calls before setting restrictions so that the PBX is not blocked by standard operations.

In addition to these settings, you can also configure similar rules for specific users or groups of users. For more information, refer to Disabling outgoing calls to certain countries or regions (page 216).

4.7.18 Tips for Apple iPad

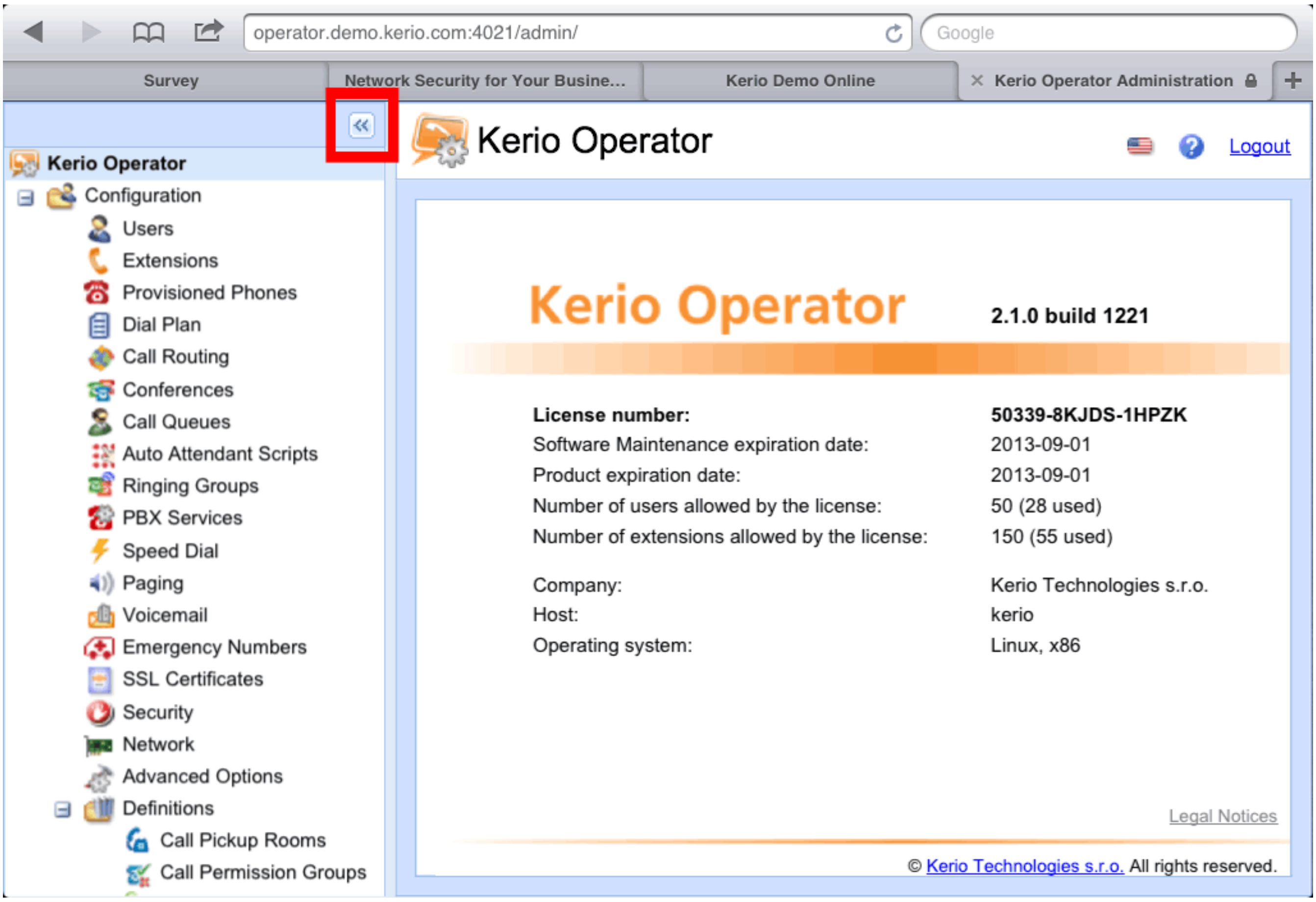
This topic provides a few useful tips for a better administration user experience on Apple iPad.

Screen orientation

It is recommended that iPad is held in the landscape mode while working with the Kerio Administration interface. For viewing longer dialog boxes, hold the device in the portrait mode.

Tree of sections

To get more space to view the section content, hide the tree of sections on the left.



Pop-up menu

To open context menu (e.g. in logs), tap the screen with two fingers at a time.

Sort by columns

Select the column and tap to set sorting or open a menu.

Editing table values

First, select a table row. To change the value, single-tap the particular spot.

Logs

- If you use search, you can go to the previous or next occurrence by using the arrow buttons.
- Log pages can be scrolled by dragging with fingers. The more fingers you use, the faster the page scrolls.

> **NOTE**
> If you have Multi-Touch allowed on iOS 5, you can use up to three fingers for log scrolling.

4.7.19 Using paging groups and services

Paging, also known as intercom or public address, enables Kerio Operator users to broadcast a message to a user or a group using a phone's speakers. Phones included in the paging group or service answer the call automatically, and activate the loud speaker.

WARNING

Paging works with phones that support auto-answer functionality.

The paging group is a group of users to whom you can make a call with using loud speaker.

The paging service is a prefix for paging. You dial the prefix + an extension to page a particular user.

Configuring paging groups

1. In the Kerio Operator administration interface, click **Paging**.
2. Click **Add Group**.
3. Type the paging group extension.
4. To add members to the group, click **Add**.
5. (Optional) Check **Page only idle extensions**. Paging does not interrupt active calls.
6. (Optional) Check **Beep when the call is established**. Your phone beeps when all phones from the paging group are connected.
7. Select audio transfer strategy: Select **only to the receiving party** to broadcast the message without giving paging group members ability to answer. Select **in both directions** to enable two-way communication.
8. (Optional) To enable call recording, select **Record Calls**.
9. Click **OK**

If you want to check your configuration, dial the group extension and do a test call.

Configuring a paging service

1. Go to the administration interface, and click **Paging**.
2. Click **Add Service**.
3. Type **Paging service prefix**.
4. (Optional) Check **Page only idle extensions**. Paging do not interrupt active calls.
5. (Optional) Check **Beep when the call is established**. Your phone beeps when all phones from the paging group are connected.
6. Decide, if you want to transfer audio **only to the receiving party** (telephones play the message and users cannot answer) or **in both directions** (telephones play the message and users can answer).
7. (Optional) To enable call recording, select **Record Calls**.
8. Click **OK**

If you want to check your configuration, dial the service prefix and an extension and do a test call.

Securing paging

Anyone who knows the extension or whole telephone number of the paging group can use this feature. You can secure your paging groups and service with **Call Permissions**. You can create a new call permission group, where paging an extension or a prefix is denied and add people without permission for using paging:

1. In the administration interface, go to **Definitions > Call Permission Groups**.
2. Click **Add**.

3. In the **Edit Call Permission Group** dialog, type a group name (for example `Paging`).

4. Click **Add**.

5. In the **Add Prefix** dialog, type a paging extension or service.

6. Click **OK**.

7. Go to **Configuration > Extensions**.

8. Select the user who will have paging disabled and click **Edit**.

9. In the **Call permissions group** menu, select the paging rule (in our example it is `Paging`).

10. Repeat step 9 to disable paging for additional users.

For testing purposes you can add yourself to restricted group called `Paging`. Try to call the paging group or service.

List of supported and tested phones

Paging was tested by Kerio Technologies with the following telephones:

- Cisco SPA508G, SPA525G
- Linksys SPA942, SPA922
- Polycom IP335, IP650
- Well SIP-T38G
- Snom 360, 820 and MeetingPoint

4.7.20 Integrating Kerio Connect and Kerio Operator

What are the possibilities of Kerio Operator and Kerio Connect integration

There are several posibilities how to integrate Kerio Operator and Kerio Connect:

Integrating voicemail

The integration synchronizes flags which marks whether a voicemail message has been read/played. If you mark a message as read in Kerio Phone or if the message is marked as read after you hear it on your phone, the message will also be flagged as read in your mailbox (and vice versa).

If integration with Kerio Connect is set, voicemail messages are not stored in Kerio Operator but in user's **Inbox** on the mailserver.

> **WARNING**
> Limitation: You can integrate Kerio Connect with a single Kerio Operator only.

Searching the address book on Kerio Connect on provisioned phones

For more information, refer to Accessing company contacts through LDAP on provisioned phones (page 177).

Calling directly from Kerio Connect Client

For more information, refer to Configuring Click to Call in Kerio Connect client (page 303).

Configuring voicemail integration

If you want to set up voicemail integration, follow these steps:

1. Go to **Configuration > Users**.

2. In the users' settings, type their email addresses.

> **WARNING**
> Use the primary email address (not an alias) — otherwise sending of messages to Inbox will not work.

3. Go to **Configuration > Voicemail > Email**.

4. Change the SMTP server settings to **Integrate with Kerio Connect**.

5. Click **Configure** and type the DNS name of Kerio Connect.

> **NOTE**
> If the IMAP service runs on a nonstandard port in Kerio Connect, enter the server name including the port number (`hostname:12345`)

6. Specify the name and password of a user with admin rights for Kerio Connect.

Authentication details are used for the first connection to Kerio Connect and creation of a special account using JSON-RPC2 API for authentication. Once this special account is created, the PBX drops the administrator's name and password.

> **NOTE**
> To synchronize flags between the two servers, Kerio Operator uses protocol IMAP with TLS or IMAPS. If Kerio Connect is behind firewall, enable at least one service on standard port. The IMAP or IMAPS services need to be allowed on Kerio Connect server.

Opening ports

If servers are behind a firewall, open the following ports:

- 143/993
- 4040

Troubleshooting

If Kerio Connect is protected by firewall, verify that the 143/993 and 4040 ports for the IMAP/IMAPS protocols are open.

The IMAP/IMAPS services must be running in Kerio Connect.

If you cannot connect Kerio Operator with Kerio Connect, consult the following logs:

- In Kerio Operator, consult the Warning log for any problems with the IMAP service.
- In Kerio Operator, consult the Error log for problems with connection to Kerio Connect's IMAP server.
- In Kerio Connect, consult the Mail log for information about delivered voicemails.

4.7.21 Configuring Click to Call in Kerio Connect client

> New in Kerio Operator 2.3!

Users of Kerio Connect client can click a contact's phone number to initiate a call from Kerio Operator. By clicking a number, you can select the registered phone/device to dial from. The selected phone/device will ring. Answer the call and Kerio Operator will place the outbound call to the dialed number.

To set up and use the Click to Call feature in Kerio Operator, go to the Configuring a number transformation section.

To set up and use the Click to Call feature in Kerio Connect, go to the Making calls from Kerio Connect topic.

If you want to Click to Call for Kerio Operator plugin for Chrome and Firefox, go to the Using Click to Call for Kerio Operator plugin for Chrome and Firefox topic.

Configuring a number transformation

Numbers dialed by Click to Call must be in the same format as for usual calls. If you use an outgoing prefix in your environment, you must add a number transformation rule to Kerio Operator:

1. In the administration interface, go to**Integration**.

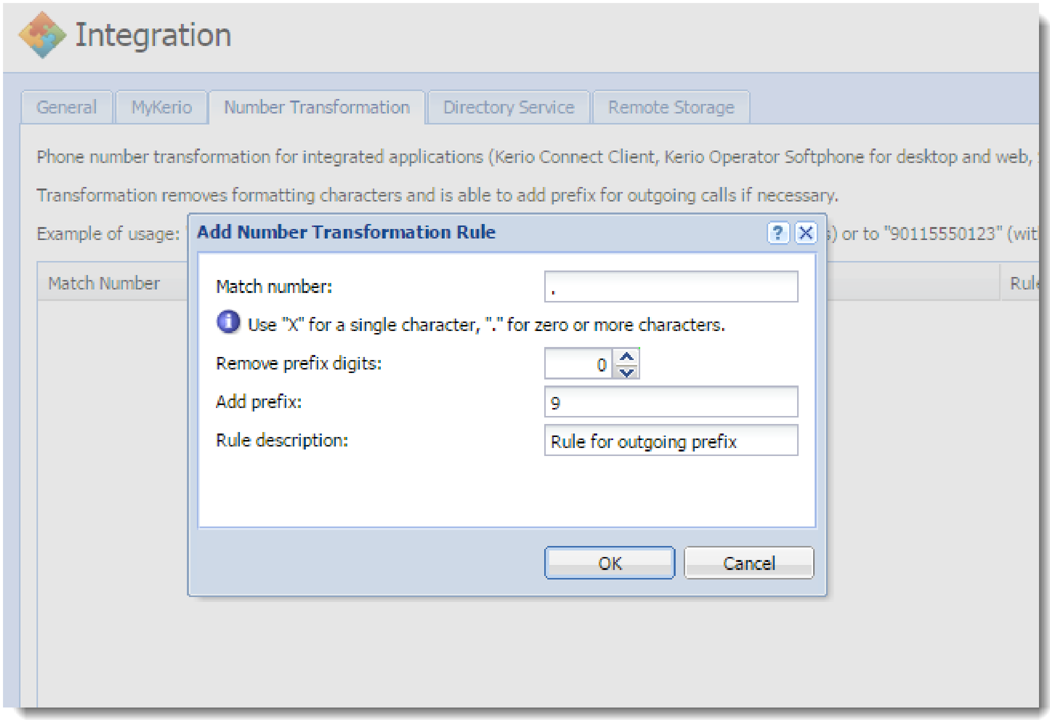
2. On the **Number Transformation** tab, add the rule for your outgoing prefix (for example 9).

3. Click **Add**.

4. In the **Add Number Transformation Rule** dialog, type dot in the **Match number** field. Numbers of any length are matched.

5. In the **Add prefix** field, add the outgoing prefix (for example 9).

6. Click **OK**



Screenshot 63: Rule for outgoing prefix

4.7.22 Configuring Remote FTP/SFTP Storage

You can configure a FTP or SFTP server as your remote storage to store data backups and call recordings.

Configuring a FTP server

1. In the administration interface, go to **Integration > Remote Storage**.

2. Key in the **Hostname** of the FTP server.

3. Key in your FTP server **Username** and **Password**.

4. Click **Test Connection**.

Configuring a SFTP server

1. In the administration interface, go to **Integration** > **Remote Storage**.

2. Key in the **Hostname** of the SFTP server.

3. Key in your SFTP server **Username**.

4. Authenticate using your preferred method as described below:

Authenticate using a password

1. Key in your SFTP server **Password**.

2. Click **Test Connection**.

Authenticate using ssh keys

1. Check **Use SFTP (SSH File Transfer Protocol)**.

2. If the public key is already configured on the SFTP server and you have a private ssh key, click **Upload Private Key**. Else, click **Generate Key Pairs** to create new keys for integration. A public key gets downloaded to be configured on the SFTP server.

> **NOTE**
> If you are replacing an existing SFTP server configuration with new a SFTP server, click **Remove Key** before uploading or generating a new pair of keys.

3. Click **Test Connection**.

5 Troubleshooting

This section helps you fix problems you might encounter when using Kerio Operator.

5.1 Common issues

This section provides solution to various issues:

5.1.1 Troubleshooting connections to SIP providers

This topic describes what information you must acquire from your provider and offers tips on configuring SIP interfaces in Kerio Operator.

The sections below cover these topics:

- SIP servers
- Domains
- User names
- Phone numbers
- SIP headers

For more information, refer to Connecting to VoIP service providers (page 25).

For more information, refer to Mapping external and internal numbers (page 196).

SIP SRV records

SRV (service) records are entries in DNS that specify the location of service servers. Some SIP providers have SIP SRV records defined for their domain name. Asterisk uses SIP SRV resolution for outbound calls.

To make outbound calls, you must add all proxy servers from your provider's SRV records to your Kerio Operator SIP interface.

To obtain proxies, you can:

- Ask your provider directly.
- Use a Linux dig command.

Example for nexvortex.com:

Command:

```
dig _sip_udp.nexvortex.com SRV
```

Result:

```
_sip._udp.nexvortex.com 1800   IN    SRV     20 0 5060 px5.nex-
vortex.com.
_sip._udp.nexvortex.com 1800   IN    SRV     30 0 5060
px7.nexvortex.com.
_sip._udp.nexvortex.com 1800   IN    SRV     10 0 5060 px1.nex-
vortex.com.
```

To add the proxy servers to the Kerio Operator SIP interface:

1. In the Kerio Operator administration interface, go to **Configuration > Call Routing**.

2. Double-click the SIP interface.

3. Go to the **SIP Details** tab.

4. Type the names of the proxy servers in the **Inbound proxy** field. `px5.nexvortex.com`, `px7.nexvortex.com`, and `px1.nexvortex.com` in the example above.

5. Click **OK**

Domains and usernames

If you have issues related to domain or user names while configuring a SIP interface:

» Verify that the Authentication usernames and the SIP name are correct. If you don't get any Authentication username from your provider, assume that they are the same.

» Verify your provider's domain name. Some providers use different terminology, for example, "server name" or "identifier to be used instead of the host name part of the SIP URI".

» Verify that the provider uses the same server name for all SIP server roles (registration, inbound proxy, outbound proxy). If not, configure the Kerio Operator SIP interface correctly for your provider.

» If your provider has multiple SIP servers, type all of them in the Kerio Operator SIP interface.

Phone numbers

Number formats

Before you configure a SIP interface and incoming and outgoing routes in Kerio Operator:

» Verify the format of phone numbers your provider uses.

- The specific number of digits, for example, 9-digit numbers, 10-digit numbers, and so on. note that some US providers use 11-digit numbers instead of 10-digit numbers. In that case, the first digit is always `1`.
- E.164 number format, where numbers start with the + sign followed by a country code (for example, `+1` in the US). This format often requires configured number rewriting for outgoing calls. For example, rewrite dialed numbers that start with `9` so that the called numbers start with `+1` (`94084964500` to `+14084964500`).
- Custom number format, where, for example, providers use the international format without the + sign. Rewrite all national and international numbers to the custom format of the provider.

» Verify that the number format for inbound calls and outbound calls is the same. If not, configure number rewriting correctly for your provider. For example, your provider sends you 10-digit numbers, but requires 11-digit numbers for

outbound calls. Create an outgoing route that rewrites numbers to 11-digit format.

Phone numbers as identifiers

Your provider can use phone numbers instead of SIP usernames. If you have only a single external number, many providers use your external number as a SIP username as well.

If you have multiple numbers, your provider:

» Uses one of the external numbers as the SIP username.

» Gives you a random string or the common part of your external numbers.

» Gives you multiple accounts, each account with a single external number. Create a SIP interface for each account. You can then assign all interfaces under the same dial-out prefix.

SIP headers

Call setup

When a device initiates a SIP call, it sends the SIP INVITE request. The beginning of the request looks like this:

```
INVITE sip:13@10.10.1.13 SIP/2.0
Via: SIP/2.0/UDP 10.10.1.99:5060;branch=z9hG4bK343bf628;rport
From: "Test 15" <sip:15@10.10.1.99>tag=as58f4201b
To: <sip:13@10.10.1.13>...
```

In the request above, extension 15 calls extension 13.

» The called number is in the INVITE (**Request-Line** in the Kerio Operator administration interface) and To headers.

» The calling number is in the From header.

After you create a SIP interface, Kerio Operator reads the calling number from the From header and the called number from the INVITE header by default. Verify which headers your provider requires and change the settings in the Kerio Operator SIP interface.

For more information, refer to Connecting to VoIP service providers (page 25).

Example

» You have multiple numbers in range 5551200-5551299.

» Your username is the common part of your numbers, 55512.

The SIP provider sends 55512 in the INVITE header and the specific number, for example 5551234, in the To header.

For outgoing calls, your provider requires 55512 in the From header and the calling number in the P-Preferred-Identity header.

Transferring calls

When transferring calls, Kerio Operator can notify the receiving party about the original caller, otherwise the callee cannot recognize the origin of the call.

Enable the **Diversion** header on the **SIP Details** tab of the SIP interface.

If the Diversion header doesn't work, ask your provider which SIP header to use.

5.1.2 Troubleshooting call quality issues

When using voice over IP, there are many considerations which can affect the quality of phone calls, and identifying the source of the issue can sometimes be difficult. This topic is designed to describe the different types of call quality issues, the general causes, and a likely solution to each type of issue.

Symptom:

One-way, or No Audio

Cause:

Improper NAT settings, or improper handling of SIP/RTP by the firewall.

Solution:

Avoid NAT, Bypass the firewall by connecting Kerio Operator directly to the WAN, and attaching a second network interface to the local network. Otherwise, verify the NAT settings in Kerio Operator and the connected SIP device. Configuration details are available in KB topic Configuring NAT. Also check the kerio forums or other resources for known SIP handling issues with the brand of firewall in front of Kerio Operator. The firewall should be configured to allow TCP/UDP port 5060 and 5061, and UDP ports 10000 ~ 20000

Symptom:

Garbled, or incomprehensible voice transmission.

Cause:

Bad Codec

Solution:

Identify and remove the bad Codec from the selected codecs in the call route.

> **NOTE**
> Troubleshooting tip: During a call, review Status > Calls and check the used Codec. Compare it to calls that don't have the issue.

Symptom:

Choppy voice quality. Some words or parts of the conversation are lost or cut off.

Causes:

- Insufficient or jittery Internet bandwidth.
- Local networking issue (bad cable, overloaded switch, bad NIC, overloaded WiFi)
- High CPU usage or insufficient hardware resources

Solutions:

- Set GSM as the only selected Codec on the call route. Research options for allocating more bandwidth or incorporating QoS.

- Update firmware of the phone, replace the cable to the phone, change the Codec of the extension, replace the phone.
- Upgrade hardware to faster storage device (e.g. SSD) or add RAM

> **NOTE**
> Troubleshooting tips: Enable extension 81 for the echo test. Dial from various locations and phones to isolate the conditions under which the choppiness occurs. From a browser on that network, go to pingtest.net and check the line quality. Check the status > system health to make sure the CPU/Memory is not overloaded. Check settings for call recording and logging to ensure reduced File I/O.

Capturing the network communication (packet capture or packet dump):

If the previously mentioned steps were not helpful, it may be necessary to capture the network communication for analysis. To do this, navigate to Configuration > Network. Select the network interface which is linked to the network where the issue is observed. Click the 'Packet Sniffer...' button to the right of the interfaces dialog. Start the capture, then attempt to reproduce the problem. Once the issue has been observed, stop the capture and download the capture file. If you have an open case with technical support, they may request this file.

5.1.3 Browser extensions or add-ons may interfere with Kerio products

When you have trouble working with an administration or client interface of Kerio products, you can try to disable or uninstall all your browser's extensions/add-ons.

Here are some tips on how to do it in the most common browsers:

- **Google Chrome** — Disable your extensions or run the browser in the incognito mode.
- **Mozilla Firefox** — Disable your add-ons or run the browser in Save Mode.
- **Safari** — Turn all extension off.
- **Internet Explorer** — Disable your add-ons or run the browser in **No Add-ons** mode.

5.1.4 Cannot play voicemails or audio files in Safari

What happens

Do you have any problem with playing voicemails or audio files in Kerio Operator administration or Kerio Phone? Playing of audio files via HTML5 is not possible when the SSL certificate of the hostname is not trusted.

Where it happens

It happens in:

- Safari on Apple Mac OS X systems,
- Safari on Apple iPhone,
- Safari on Apple iPad,
- Safari on Windows.

How to fix it

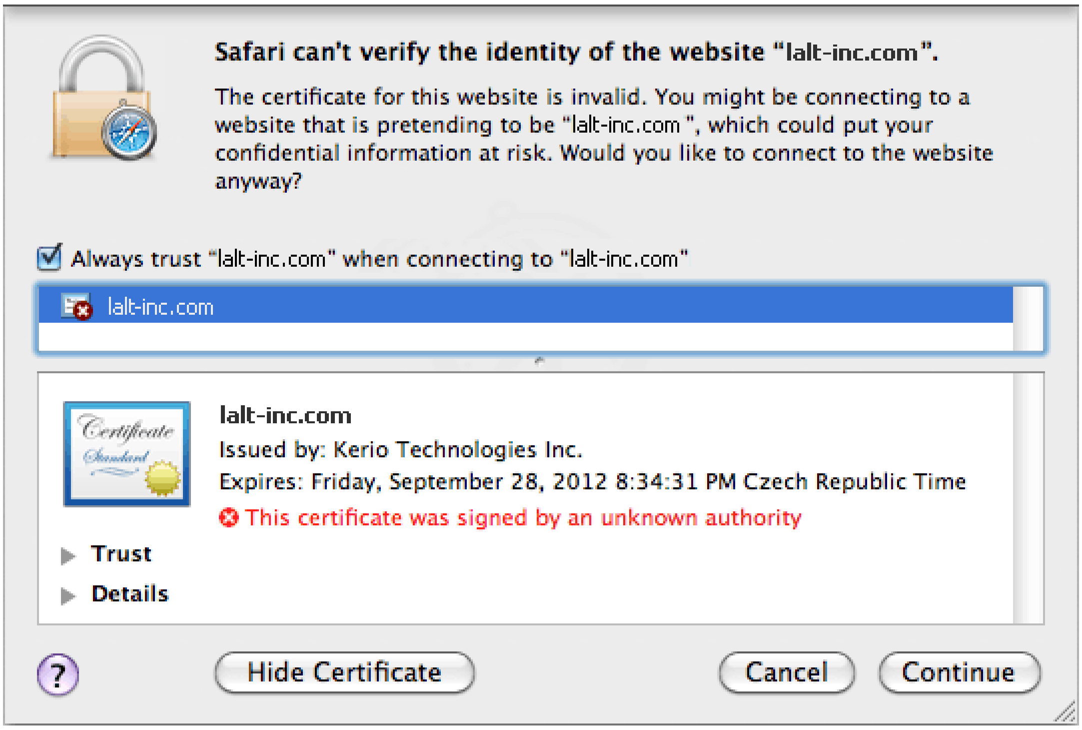
Use a certificate verified by a Certification Authority, or install a self-signed SSL certificate properly and mark it as trusted on your device.

WARNING

In Safari on Windows you need a certificate verified by a Certification Authority. Self-signed certificates do not work.

How to set self-signed certificate as trusted on Mac OS X systems

1. Type your Kerio Operator address to the Safari browser. SSL certificate warning appears.
2. Click **Show Certificate**.
3. Check **Always trust**.



4. Click **Continue**.
5. Enter administrator's username and password for authentication.

How to set self-signed certificate as trusted on Apple iPhone and Apple iPad

1. Type your Kerio Phone address to browser. The login window appears.
2. Tap on the **Download SSL certificate** link.
3. Install the certificate.

NOTE

This certificate will also be used for access to the administration interface too.

5.2 Vulnerabilities

Vulnerability	Description
Bash vulnerability CVE-2014-6271, CVE-2014-7169 (ShellShock)	The shellshock vulnerability (aka CVE-2014-6271 and CVE-2014-7169) is a security bug affecting Unix-like operating systems through the Bash shell. For information on its impact on Kerio products, read Bash vulnerability CVE-2014-6271, CVE-2014-7169 (ShellShock) article.
Linux Glibc vulnerability CVE-2015-7547	A vulnerability in the Linux glibc system library has been found. An attacker can gain root access to the server and execute a code. For more details on its impact on Kerio products, read Linux Glibc vulnerability CVE-2015-7547 article.
Linux vulnerability CVE-2015-0235 (GHOST)	There is a vulnerability in Linux glibc system library. An attacker can exploit this vulnerability and gain root access to your server and execute a code. For more details on its impact on Kerio products, read Linux vulnerability CVE-2015-0235 (GHOST) article.
OpenSSL vulnerability CVE-2014-0160 (Heartbleed)	The National Institute of Standards and Technology (NIST) has published a vulnerability to OpenSSL 1.0.1. Details regarding the vulnerability are available from the NIST website. Kerio Operator 2.2.0 up to 2.2.4 used the affected version of the OpenSSL library. However, a fix is available for Kerio Operator as of version 2.2.5. You can download this release from the Kerio Website. For additional information and security precautions, read OpenSSL vulnerability CVE-2014-0160 article.
SSL 3.0 vulnerability CVE-2014-3566 and POODLE	This vulnerability is a flaw in the protocol design. An attacker that controls the network between the client and the server can interfere with any attempted handshake offering TLS 1.0 or later and force both client and server to use SSL 3.0 protocol instead. They can then use other attack techniques (eg. BEAST attack) to decipher transmitted data. For information on its impact on Kerio products, read SSL 3.0 vulnerability CVE-2014-3566 (POODLE) article.

5.3 USB Tools

This section provides information about password recovery, factory reset, and diagnosing Kerio Operator hardware appliances via an USB flash drive.

5.3.1 Restoring the Kerio Operator default configuration using a USB flash-drive

Kerio Technologies provides a set of tools for solutions for situations in which it is not possible to connect to Kerio Operator on a network and administer it through the Kerio Operator Administration web interface.

These tools are designed to run from a USB flash-drive.

For complete system recovery a USB flash-drive with capacity of at least 1 GB is required. For restoring the default configuration, a capacity of 256 MB is needed.

Should any issues arise (for example, if Kerio Operator fails to work even after you perform a complete system recovery) please contact our technical support.

Restoring default configuration

The factory settings of Kerio Operator can be recovered with the file kerio-operator-factory-reset.

Factory settings recovery includes removal of all configuration data including activation and the statistics database.

This USB tools is designed for a single use so that an operation will not repeat if you restart with the flash-drive still in the USB port. Once you perform the operation, the content cannot be reused, so the file can be removed from the flash-drive.

1. Insert a USB flash-drive to your computer (256 MB or larger) into a USB port on your computer.

2. Make sure that only one partition with file system FAT16 or FAT32 (VFAT) is created on the flash-drive. The USB drive must not be formatted by file system NTFS or ext2, ext3, or ext4.

3. Save the kerio-operator-factory-reset file to the flash-drive.

4. Switch off Kerio Operator.

5. Plug the USB flash-drive into one of the USB ports of your Kerio Operator.

6. Switch on Kerio Operator.

7. For factory settings recovery to take effect, Kerio Operator is restarted automatically.

8. To connect to Kerio Operator, set the following TCP/IP parameters on your computer:

- IP address: `10.10.10.2`
- Subnet mask: `255.255.255.0`

9. Use the web browser of the connected computer to enter the following address:
`https://10.10.10.1:4021/admin`

10. Set the administrator password, login to the product administration and configure **Kerio Operator Box** as needed.

> **WARNING**
> If the steps above do not work, try another flash-drive. Different Kerio Operator Box models require different USB drive formats:

» Kerio Operator Box 1210, 3210 and 3230 require a USB flash-drive formatted like a floppy disk (not partitioned).

» Kerio Operator Box 1220 requires a flash-drive formatted with a master boot record (MBR). USB drives with floppy-type formatting cannot connect to Kerio Operator Box, but can be reconfigured to work. See Formatting USB flash-drive with MBR below.

Complete system recovery

Kerio Operator can be completely recovered with the kerio-operator-rescue file. Within the system recovery, all configuration data including activation and the statistics database will be completely rewritten. Therefore the device will have to be reactivated and reconfigured for further use.

> Before applying complete system recovery, it is highly recommended to retest connection to Kerio Operator after attempting for restore of the factory settings.

Preparing flash-drive for system recovery

For complete system recovery, Kerio Operator Box first needs to introduce operating system from USB drive. File kerio-operator-rescue is an image of an installation drive and must be saved directly on the physical device (similarly as in case of burning ISO images on CD). Please follow the instructions according to your client system.

Microsoft Windows

1. Mount the USB flash-drive to your computer. If necessary, back up files saved on the drive. The flash-drive data will be rewritten completely!

2. Download and unpack Image Writer (it does not require installation).

3. Download file kerio-operator-rescue.

4. In application **Image Writer**, look up this file, select your flash-drive and click on **Write**.

5. Remove the drive securely and unplug it from your computer.

Linux

1. Mount the USB flash-drive to your computer. If necessary, back up files saved on the drive. The flash-drive data will be rewritten completely!

2. Download file kerio-operator-rescue.

3. Run the terminal (console).

4. Use command `sudo fdisk -l` to detect the USB flash-drive name (e.g. `/dev/sdb`).

5. Save the kerio-operator-rescue file to the USB flash-drive using command: `sudo dd if=rescue.img of=/dev/sdx bs=1M` and replace `rescue.img` with the real file name and `/dev/diskX` with the real appliance. It is necessary to enter the physical device (e.g. `/dev/sdx`), not only a partition (e.g. `/dev/sdx1`).

6. Use command `sudo sync` to guarantee finishing of all drive operations.

7. Unplug the USB drive from your computer.

Mac OS X

1. Mount the USB flash-drive to your computer. If necessary, back up files saved on the drive. The flash-drive data will be rewritten completely!

2. Download file kerio-operator-rescue.

3. Run the terminal (**Applications > Utilities > Terminal**).

4. Use command `sudo diskutil list` to detect the USB flash-drive name (e.g. `/dev/DiskX`).

5. Use command `sudo diskutil unmountDisk /dev/diskX` to unmount the drive.

6. Save file kerio-operator-rescue to the USB flash-drive by using command: **sudo dd if=rescue.img of=/dev/Disk1 bs=1m** and replace `rescue.img` with the real file name and `/dev/diskX` with the real appliance.

7. Unplug the USB drive from your computer.

Kerio Operator Box device system recovery

1. Switch off Kerio Operator Box.

2. Plug the USB flash-drive into one of the USB ports of your **Kerio Operator Box**.

3. Start the Kerio Operator Box and wait for a sound signal.

4. To connect to Kerio Operator, set the following TCP/IP parameters on your computer:

- IP address: `10.10.10.2`
- Subnet mask: `255.255.255.0`

5. Use the web browser of the connected computer to enter the following address: `https://10.10.10.1:4021/admin`

6. Set the password, login to the product administration and configure **Kerio Operator Box** as needed.

Recovering USB flash-drive for further use

Special partitions are now created on the USB flash-drive and part of the space is unused. To reuse the drive again as an external desk for other purposes, remove all drive partitions, create one or more new partitions and reformat the drive by an appropriate file system.

Please follow the instructions according to your client system.

Microsoft Windows

1. Run the **Command Line**.

2. Enter command `diskpart`. On Windows Vista and Windows 7 confirmation of running the application under administration account can be required.

3. Use command `list disk` to show the list and look up the number of the physical disk.

4. Enter command `select disk 8` (replace number 8 by the number of the corresponding disk).

5. Use command `clean` to remove all created partitions.

6. Create a new disk partition by using the following commands, as listed:

```
create partition primary
select partition 1
format fs=fat32 label="USB Flash"
exit
```

Linux

Use graphical tool GParted or command `fdisk`.

Mac OS X

Use system tool Disk Utility (**Application > Utilities > Disk Utility**).

Formatting USB flash-drive with MBR

1. Connect the USB flash-drive to a computer with Windows operating system.

2. Run the Command Line.

3. Enter command `diskpart`. On Windows Vista and Windows 7 confirmation of running the application under administration account can be required.

4. Use command `list disk` to show the list and look up the number of the physical disk.

5. Enter command `select disk 8` (replace number 8 by the number of the corresponding disk).

6. Use command `clean` to remove all created partitions.

7. Create a new disk partition by using the following commands, as listed:

```
create partition primary
select partition 1
format fs=fat32 label="USB Flash" quick
exit
```

5.3.2 Restoring the Kerio Operator Box V series default configuration using a USB flash drive

Kerio Technologies provides a set of tools for solutions for situations in which it is not possible to connect to Kerio Operator on a network and administer it through the Kerio Operator Administration web interface.

You can upgrade the system in your Kerio Operator V series box with a USB flash drive.

You need a flash drive with the capacity of at least 1 GB to run the tools. For restoring the default configuration, 256 MB suffice.

If you have any issues after using the tools, for example, if Kerio Operator fails to work even after you perform a complete system recovery, please contact our technical support.

Restoring the default configuration

Recovering to factory settings includes removal of all configuration data including activation and the statistics database.

This USB tool is designed for a single use so that the operation does not repeat if you restart with the flash drive still in the USB port. You can remove the files from the flash drive after the upgrade.

1. Insert the flash drive into a USB port on your computer (256 MB an more).

2. Make sure that only one partition with the **FAT16** or **FAT32 (VFAT)** file system is created on the flash drive. The USB disk must **not** be formatted by the NTFS, ext2, ext3, or ext4 file systems.

3. Download and save the kerio-operator-factory-reset file to the flash drive.

4. Switch off Kerio Operator.

5. Plug the USB flash drive into a USB port of your Kerio Operator box.

6. Switch on the Kerio Operator box.

7. Kerio Operator restarts automatically.

8. To connect to Kerio Operator, set the following TCP/IP parameters on your computer:

- IP address: 10.10.10.2
- Subnet mask: 255.255.255.0

9. In your web browser, type this URL: `https://10.10.10.1:4021/admin`

10. Activate the product, login to the product administration interface and configure Kerio Operator as needed.

Running a complete system recovery

During the complete system recovery, all configuration data, including activation and the statistics database, is completely rewritten. This means that you must reactivate and reconfigure the device afterwards.

> **IMPORTANT**
> Before doing a complete system recovery, restore the default configuration and then retest the connection to Kerio Operator.

Preparing a flash drive for a complete system recovery

For a complete system recovery, you must save the installation disk image directly to the physical device.

Microsoft Windows

1. Insert the flash drive into a USB port on your computer (1 GB an more).

> **NOTE**
> All data on the flash drive will be completely overwritten.

2. Download and unpack Image Writer (it does not require installation).

3. Download the kerio-operator-rescue file.

4. In Image Writer, find the file, select your flash drive and click **Write**.

5. Eject the flash drive securely and remove it from your computer.

Linux

1. Insert the flash drive into a USB port on your computer (1 GB an more).

> **NOTE**
> All data on the flash drive will be completely overwritten.

2. Download the kerio-operator-rescue file.

3. Run the terminal (console) in the super-user mode (for example, using the `su` or `sudo -s` command depending on your Linux distribution).

4. Use the command `fdisk -l` to detect the USB flash drive name (for example, `/dev/sdx`).

5. Save the kerio-operator-rescue file to the flash drive using the following command: `dd if=rescue.img of=/dev/sdx bs=1M` and replace `rescue.img` with the real file name and `/dev/sdx` with the actual device name. You must type the physical device (for example, `/dev/sdx`), not a partition (for example, `/dev/sdx1`).

6. Use the `sync` command to ensure all disk operations finish.

7. Eject the USB drive safely and remove it from the USB port.

Mac OS X

1. Insert the flash drive into a USB port on your computer (1 GB an more).

> **NOTE**
> All data on the flash drive will be completely overwritten.

2. Download the kerio-operator-rescue file.

3. Run the terminal: **Applications > Utilities > Terminal**.

4. Use the `sudo diskutil list` command to detect the USB flash drive name (for example, `/dev/diskX`).

> **NOTE**
> The drive name is case sensitive.

5. Use the `sudo diskutil unmountDisk /dev/diskX` command to unmount the flash drive.

6. Save the kerio-operator-rescue file to the USB flash drive using the following command: `sudo dd if=rescue.img of=/dev/disk1 bs=1m` and replace `rescue.img` with the real file name and `/dev/diskX` with the actual device name.

7. Eject the flash drive securely and remove it from your computer.

Kerio Operator Box system recovery

1. Switch off Kerio Operator.

2. Plug the USB flash drive into a USB port of your Kerio Operator Box.

3. Switch on Kerio Operator.

4. To connect to Kerio Operator, set the following TCP/IP parameters on your computer:

- IP address: 10.10.10.2
- Subnet mask: 255.255.255.0

5. In your web browser, type this URL: `https://10.10.10.1:4021/admin`

6. Activate the product, login to the product administration interface and configure Kerio Operator as needed.

Recovering the USB flash drive for further use

The recovery file creates partitions on the USB flash drive. To reuse the USB drive for other purposes, you need to remove all disk partitions, create new partitions, and reformat the disk for your file system.

Microsoft Windows

1. Click **Start** and in the **Search** field type `cmd.exe` to open the **Command Prompt** window.

2. In the command line, type `diskpart`. You may need to confirm that you have administration rights.

3. Type `list disk` to display the list of drives and look up the number of the physical USB drive.

4. Type `select disk X` (replace `X` with the number of the corresponding disk).

5. Type `clean` to remove all partitions.

6. Create a new disk partition by typing the following commands in the order listed:

```
create partition primary
select partition 1
format fs=fat32 label="USB Flash" quick
exit
```

Linux

Use the graphical tool GParted or the command `fdisk`.

Mac OS X

Use the system tool Disk Utility: **Application > Utilities > Disk Utility**.

Formatting a USB flash drive with MBR

1. Click **Start** and in the **Search** field type `cmd.exe` to open the **Command Prompt** window.

2. In the command line, type `diskpart`. You may need to confirm that you have administration rights.

3. Type `list disk` to display the list of drives and look up the number of the physical USB drive.

4. Type `select disk X` (replace X with the number of the corresponding disk).

5. Type `clean` to remove all partitions.

6. Create a new disk partition by typing the following commands in the order listed:

```
create partition primary
select partition 1
format fs=fat32 label="USB Flash" quick
exit
```

Related articles

For more information, refer to Recovering your Kerio Operator Box V series password using a USB flash drive (page 324).

5.3.3 Diagnostic tool for Kerio Operator Box

Kerio Technologies provides diagnostic tool for diagnose hardware problems with the Kerio Operator Box, which elicits crucial information for the Kerio Technologies technical support. The tool is designed for use from a USB flashdisk.

You need a USB flasdisk with capacity of at least 256 MB.

The diagnostics tool is designed for a single use to avoid unexpected repetition of the operation upon the next restart in case that the flashdisk has not been dismounted. This implies that once you perform the operation, the disk content cannot be used again and the files can be removed.

Using a diagnostic USB tool

> **NOTE**
> Kerio Operator uses the same diagnostic tool as Kerio Control. For diagnostics, use the `kerio-control-usbdiag` file.

If you need the USB diagnostic tool, download and use the diagnostic tool.

Creating diagnostic flashdisk

The diagnostic tool is an image of an installation disk and must be saved directly on the physical device (similarly as in case of burning ISO images on CD). Please follow the instructions according to your client system.

Microsoft Windows

1. Mount the USB flashdisk to your computer. If necessary, back up files saved on the disk. The flashdisk data will be rewritten completely!

2. Download and unpack Image Writer (it does not require installation).

3. Download file the diagnostic tool file.

4. In application **Image Writer**, look up this file, select your flashdisk and click on **Write**.

5. Remove the disk securely and unplug it from your computer.

Linux

1. Mount the USB flashdisk to your computer. If necessary, back up files saved on the disk. The flashdisk data will be rewritten completely!

2. Download the diagnostic tool file.

3. Run the terminal (console).

4. Use command `sudo fdisk -l` to detect the USB flashdisk name (e.g. `/dev/sdb`).

5. Save the diagnostic tool file on this device by using the following command:

```
sudo dd if=usbdiag.img of=/dev/sdx bs=1M
```

Replace string with:

`usbdiag.img` by the real file name and `/dev/sdx` by the real device. It is necessary to enter the physical device (e.g. `/dev/sdx`), not only a fragment (for example, `/dev/sdx1`).

6. Use command `sudo sync` to guarantee finishing of all disk operations.

7. Unplug the USB disk from your computer.

Mac OS X

1. Mount the USB flashdisk to your computer. If necessary, back up files saved on the disk. The flashdisk data will be rewritten completely!

2. Download the diagnostic tool file.

3. Run the terminal (**Applications > Utilities > Terminal**).

4. Use command `sudo diskutil list` to detect the USB flashdisk name (e.g. `/dev/diskX` or `/dev/DiskY` — watch the letter case).

5. Use command `sudo diskutil unmountDisk /dev/diskX` to unmount the disk.

6. Save the diagnostic tool file on the USB disk by using the following command:

```
sudo dd if=usbdiag.img of=/dev/disk1 bs=1m
```

Replace string with:

`usbdiag.img` by the real file name and `/dev/diskX` by the real device.

7. Unplug the USB disk from your computer.

Using diagnostic flashdisk

1. Switch off Kerio Operator Box.

2. Plug the USB flashdisk into one of the USB ports of your Kerio Operator Box.

3. Switch on Kerio Operator Box.

4. Approximately after two minutes Kerio Operator Box beeps three times. This means that the operating system has been introduced and the diagnostic test has just been started. If the device does not beep within the following 10 minutes, the test has failed. In such case switch off the device, unplug the USB flashdisk and send diagnostic information the

Kerio Technologies
technical support (see below).

5. The test starts with 10 beeps and runs for about 60 minutes — a 40-minute memory test and a diagnostic test. If you want to skip the memory test, press any key during the ten-beep interval. Once the test is finished, Kerio Control Box starts beeping every 30 seconds.

6. Switch off Kerio Operator Box and unplug the USB flashdisk.

Test results processing

Plug the USB flashdisk to your computer again. There is a partition called

KerioDiag
on the disk. This partition includes the file with test results.

Please send this file to the Kerio Technologies technical support and possibly provide a description of the non-standard behavior of your Kerio Operator Box.

Recovering USB flashdisk for further use

For more information, refer to Recovering USB flash-drive for further use (page 315).

5.3.4 Diagnostic tool for Kerio Operator Box V series

Kerio Technologies provides a tool for diagnosing hardware problems with the Kerio Operator Box NG series. This tool collects crucial information for the Kerio Technologies technical support. It is designed to be run from a USB flash drive.

You need a USB flash drive with a capacity of at least 256 MB.

The diagnostic tool is designed for a single use so that the operation will not repeat if you restart with the flash drive still in the USB port. Once you perform the operation, the content cannot be reused, so the files can be removed from the flash drive.

Creating the diagnostic flash drive

> **NOTE**
> Kerio Operator uses the same diagnostic tool as Kerio Control. For diagnostics, use the `kerio-control-usbdiag` file.

The diagnostic tool is an image of an installation disk and must be saved directly on the physical device. Follow the instructions for your client system below.

Microsoft Windows

1. Insert the flash drive to your computer (256 MB or larger) into a USB port on your computer.

> **NOTE**
> All data on the flash drive will be completely overwritten, so be sure to save any files you need elsewhere.

2. Download and unpack Image Writer (it does not require installation).

3. Download the diagnostic tool file.

4. In **Image Writer**, find the file, select your flash drive and click **Write**.

5. Eject the flash drive securely and remove it from your computer.

Linux

1. Insert the flash drive into a USB port on your computer.

NOTE
All data on the flash drive will be completely overwritten, so be sure to save any files you need elsewhere.

2. Download the diagnostic tool file.

3. Run the terminal (console).

4. Use the command `sudo fdisk -l` to detect the USB flash drive name (e.g., `/dev/sdx`).

5. Save the diagnostic tool file on the flash drive using this command: `sudo dd if=usbdiag.img of=/dev/sdx bs=1M` (replace `usbdiag.img` with the real file name and `/dev/sdx` with the actual device name). You must enter the physical device (e.g., `/dev/sdx`), not the partition (e.g., `/dev/sdx1`).

6. Use the command `sudo sync` to ensure that all disk operations finish.

7. Eject the flash drive securely and remove it from your computer.

Mac OS X

1. Insert the flash drive into a USB port on your computer.

NOTE
All data on the flash drive will be completely overwritten, so be sure to save any files you need elsewhere.

2. Download the diagnostic tool file.

3. Run the terminal: **Applications > Utilities > Terminal**.

4. Use the command `sudo diskutil list` to detect the USB flash drive name (e.g., `/dev/diskX` or `/dev/DiskY`).

NOTE
This is case sensitive.

5. Use the command `sudo diskutil unmountDisk /dev/diskX` to eject the flash drive.

6. Save the diagnostic tool file on the USB flash drive using this command: `sudo dd if=usbdiag.img of=/dev/disk1 bs=1m` (Replace string `usbdiag.img` with the real file name and `/dev/diskX` with the real device).

7. Eject the flash drive securely and remove it from your computer.

Using the diagnostic flash drive

1. Switch off Kerio Operator Box.

2. Plug the USB flash drive into one of the USB ports of your **Kerio Operator Box**.

3. Switch on Kerio Operator Box to run the diagnostic test. The diagnostic test may take some time (approximately 60 minutes).

4. Switch off Kerio Operator Box and eject the USB flash drive.

Test results processing

Reinsert the flash drive into the USB port.

Find the partition called **KerioDiag** on the flash drive. It contains the file with test results.

Send this file to Kerio Technologies technical support, and optionally provide a description of the nonstandard behavior of your Kerio Operator Box.

Recovering USB flash drive for further use

To reuse your flash drive, you will need to reformat it to remove the partitions. For more information, refer to Restoring the Kerio Operator Box V series default configuration using a USB flash drive (page 316).

Related articles

For more information, refer to Recovering your Kerio Operator Box V series password using a USB flash drive (page 324).

For more information, refer to Restoring the Kerio Operator Box V series default configuration using a USB flash drive (page 316).

5.3.5 Recovering password using USB flash-drive for Kerio Operator

Kerio Technologies provides a tool for password recovery. The tool is designed for use from a USB flash-drive.

For password recovery a USB flash-drive with capacity of at least 256 MB will do.

The password recovery tool is designed for a single use to avoid unexpected repetition of the operation upon the next restart in case that the flash-drive has not been dismounted. This implies that once you perform the operation, the disk content cannot be used again and the files can be removed.

Creating and using a password recovery tool

Forgotten administration password can be recovered by using file kerio-operator-password-reset.

Follow these instructions:

1. Mount the USB flash-drive to your computer.
2. Make sure that only one fragment with file system **FAT16** or **FAT32** (**VFAT**) is created on the flash-drive. The USB disk must not be formatted by file system **NTFS** or **ext2 / ext3 / ext4**.
3. Save file kerio-operator-password-reset to the flash-drive.
4. Switch off the Kerio Operator.
5. Plug the USB flash-drive into the USB port.
6. Switch on Kerio Operator.
7. In your web browser, open the Kerio Operator Administration.
8. Activation wizard opens in the browser. As the product has already been activated, the wizard will require a new administration password.

Now you can login as user `Admin` with a new password.

WARNING

If it does not work, try another USB flash-drive.

Details of the known issue: There are two formats of USB flash-drives (the first type uses MBR as a boot sector, the second one is formatted as a floppy). Each format is eligible for different type of hardware device:

» Kerio Operator Box 1210, 3210 and 3230 — if the USB flash-drive is formatted as a floppy. The second type which uses MBR cannot connect to Kerio Operator Box.

» Kerio Operator Box 1220 — if the USB flash-drive uses MBR as a boot record, you can use it directly. The second type which is formatted as a floppy cannot connect to Kerio Operator Box. If you want to use it, format the USB flash-drive according to the following steps: Formatting USB flash-drive with MBR.

5.3.6 Recovering your Kerio Operator Box V series password using a USB flash drive

You can recover your password for the administration interface with a USB flash drive.

You need a flash drive with the capacity of at least 256 MB.

The password recovery tool is designed for a single use so that the operation does not repeat if you restart with the flash drive still in the USB port. You can remove the files from the flash drive after the upgrade.

Creating and using a password recovery tool

To recover a lost administration password:

1. Insert the flash drive into a USB port on your computer.

2. Make sure that only one partition with the **FAT16** or **FAT32 (VFAT)** file system is created on the flash drive. The USB disk must **not** be formatted by the NTFS, ext2, ext3, or ext4 file systems.

3. Download and save the file kerio-operator-password-reset to the flash drive.

4. Switch off Kerio Operator.

5. Plug the USB flash drive into a USB port of your Kerio Operator box.

6. Switch on Kerio Operator.

7. When the Kerio Operator Engine starts up, open the Kerio Operator administration interface in a browser. The activation wizard opens.

8. In the activation wizard, create a new password for the admin account.

Related articles

For more information, refer to Restoring the Kerio Operator Box V series default configuration using a USB flash drive (page 316).

6 Glossary

A

Analog telephone adapter
A device for connecting analog devices to a digital or voice over IP network.

ATA
Analog telephone adapter - A device for connecting analog devices to a digital or voice over IP network.

B

Basic Rate Interface
An ISDN channel intended for small systems which achieves up to 128kbps data rate.

BLF
Busy Lamp Field - A set of indicators that monitor the current state (online, offline, busy, on a call) of a phone extension.

BRI
Basic Rate Interface - An ISDN channel meant for small enterprise systems to obtain upto 128kbps data rate.

Busy Lamp Field
A set of indicators that monitor the current state (online, offline, busy, on a call) of a phone extension.

C

Cacti
Monitoring tool based on SNMP.

call routing
A process for routing of incoming and outgoing calls between internal extensions, PBX, and for example PSTN.

Caller ID
A service that provides information about caller's number.

Click to dial
An action that requests a real time connection via phone call.

Codecs
Programs used in streaming media and audio/video conferencing that encodes or decodes digital data streams and signals.

CPE
Customer-premises equipment - Provider's devices that are physically located on the customer's premises.

CRM

Customer Relationship Management - Strategies and technologies for managing and analyzing customer relationships.

Customer-premises equipment

Provider's devices that are physically located on the customer's premises and are connected to a telecommunication channel of the provider.

Customer Relationship Management

Strategies and technologies for managing and analyzing customer relationships.

D

DECT

Digital Enhanced Cordless Telecommunications - Cordless telephone systems.

DHCP

Dynamic Host Configuration Protocol - A protocol that automatically gives IP addresses and additional configuration to hosts in a network.

Digital Enhanced Cordless Telecommunications

Cordless telephone systems.

DTMF

Dual-tone multi-frequency signal - Tone generated by the telephone or fax device when dialling while communicating with the telephone line provider.

Dual Tone - Multi Frequency

The tone generated by the telephone (or fax) device when dialling. This is used for communicating with the telephone line provider.

Dynamic Host Configuration Protocol

A protocol that automatically gives IP addresses and additional configuration to hosts in a network.

E

Euro-ISDN

An Integrated Services Digital Network (ISDN) standards as developed by European Telecommunications Standards Institute (ETSI).

F

File Transfer Protocol

A protocol for tranferring computer files from a server.

FTP

File Transfer Protocol - A protocol for tranfering computer files from a server.

H

hardware appliance

Kerio Operator installed and delivered with standardized and tested hardware box.

I

Internet telephony service provider

A type of a service provider that provides communication via Internet. The communication is based on Voice over Internet Protocol (VoIP).

IP PBX

A Private Branch Exchange system that connects telephone extensions through internet and provides additional audio and video communication features.

ISDN

Integrated services digital network - A technology enabling digital transmission of data and voice signals over a telephone network.

ITSP

Internet telephony service provider - A type of a service provider that provides communication based on Voice over Internet Protocol (VoIP).

K

Kerio Operator App for Salesforce

An application based on Call Center that integrates Kerio Operator and Salesforce.

Kerio Operator Softphone

A softphone app for Android or iOS.

Kerio Phone

A softphone available as a native desktop or web-based application.

L

LDAP

Lightweight Directory Access Protocol enables users to access centrally managed contacts.

Lightweight Directory Access Protocol

Lightweight Directory Access Protocol enables users to access centrally managed contacts.

M

Master Boot Record

A type of a boot sector.

MBR

Master Boot Record - A type of a boot sector.

MyKerio

A web-based application for monitoring and managing appliances of Kerio products.

N

NAT

Network address translation - A method that remaps IP addresses by changing network address information.

Network address translation

A method that remaps IP addresses by changing network address information.

P

PBX

Private Branch Exchange - System that connects telephone extensions and switches calls.

prefix

Country codes, area codes, a number, or a set of numbers that are dialed before the telephone numbers.

PRI

Primary Rate Interface - An Integrated Services Digital Network channel for large enterprise systems to obtain higher speed than Basic Rate Interface.

Primary Rate Interface

An ISDN channel meant for large enterprise systems to obtain higher data transfer rate than Basic Rate Interface.

PSTN

Public switched telephone network - A global telecommunications network that operates the traditional telephony system.

Public switched telephone network

A global telecommunications network that operates the traditional telephony system.

Q

QoS

Quality of service - Network's ability to obtain maximum bandwidth and manage other network performance elements like latency, error rate and uptime.

S

Secure Sockets Layer

A protocol that ensures integral and secure communication between networks.

Service record
: Service record is a record in DNS that specifies the location of server for individual services.

Session Initiation Protocol
: A communication protocol used for voice and video calls in Internet telephony or private IP telephone systems.

Simple Network Management Protocol
: A protocol for gathering and organizing information about devices in IP networks, and changing devices behavior.

SIP
: Session Initiation Protocol - A communication protocol used for voice and video calls in Internet telephony or private IP telephone systems.

SIP interface
: An external interface used for connecting to SIP providers.

SIP password
: A password for authenticating provided by a SIP provider.

SIP Provider
: A telecommunications company that provide telephony services based on Voice over Internet Protocol.

SIP trunk
: Wide range of external numbers provided by a SIP provider.

SIP username
: An username for authenticating provided by a SIP provider.

SMTP
: Simple Mail Transport Protocol - An internet standard used for email transmission across IP networks.

SNMP
: Simple Network Management Protocol - A protocol for gathering and organizing information about devices in IP networks, and changing devices behavior.

Software Appliance
: A special operating system designed to be installed on a computer.

SRV record
: Service record is a record in DNS that specifies the location of server for individual services.

SSL
: Secure Sockets Layer - A protocol that ensures integral and secure communication between networks.

T

Telecommunications service provider

A type of a service provider that provides telephone and other services.

TFTP

Trivial File Transfer Protocol - A simple protocol for transferring files.

Trivial File Transfer Protocol

A simple protocol for transferring files.

TSP

Telecommunications service provider - A type of a service provider that provides telephone and other services.

U

UDP

User Datagram Protocol - Ensures packet transmission.

URL

The Uniform Resource Locator is the address of a web page on the world wide web.

User Datagram Protocol

Ensures packet transmission.

V

Virtual Appliance

Pre-configured Kerio Operator virtual machine image for VMware.

Voice over Internet protocol

A digital telephone system that uses the internet as the transmission medium, rather than the PSTN.

VoIP

Voice over Internet protocol - A digital telephone system that uses the internet as the transmission medium, rather than the PSTN.

VoIP Phone

A device that works on principles of VoIP.

W

webRTC

Set of communication protocols enabling real-time communication over peer-to-peer connections.

7 Legal Notices

7.1 Trademarks and registered trademarks

Aastra® is registered trademark of Aastra Technologies Limited.

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Cisco® and Linksys® are registered trademarks of Cisco Systems, Inc.

Digium® is registered trademark of Digium, Inc.

Firefox® is registered trademark of Mozilla Foundation.

Grandstream® is a registered trademark of Grandstream Networks, Inc.

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Polycom® is registered trademark of Polycom, Inc.

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SJphone® is registered trademark of SJ Labs, Inc.

snom® is registered trademark of snom technology AG.

snom® is registered trademark of snom technology AG.

Wireshark® is registered trademark of Wireshark Foundation.

X-Lite is a software phone developed by CounterPath Corporation with registered trademark of CounterPath®.

7.2 Used open source software

This product contains the following open-source libraries:

adapter.js

Shim to insulate apps from spec changes and prefix differences.

Appliance OS Sources

Kerio Operator devices are based on open software from various resources. For detailed information on conditions of each particular software used in the product, refer to acknowledgments.

To download the source package, go to http://download.kerio.com/archive/.

asterisk

Asterisk - An open source telephony toolkit.

AudioContext-Polyfill

Polyfill for AudioContext and its parties on Web Audio API.

coturn

coturn TURN server project

Heimdal Kerberos

Heimdal is an implementation of Kerberos 5, largely written in Sweden. It is freely available under a three clause BSD style license (but note that the tar balls include parts of Eric Young's libdes, which has a different license). Other free implementations include the one from MIT, and Shishi. Also Microsoft Windows and Sun's Java come with implementations of Kerberos.

jsonrpccpp

C++ framework for json-rpc (json remote procedure call)

Kerio Asterisk Module

The Kerio Asterisk Module extends the functionality of the Asterisk PBX to match Kerio Operator needs. It is distributed and licensed under GNU General Public License version 2. The complete source code is available at:

http://download.kerio.com/archive/

libcurl

Libcurl is a free and easy-to-use client-side URL transfer library. This library supports the following protocols: FTP, FTPS, HTTP, HTTPS, GOPHER, TELNET, DICT, FILE and LDAP.

libiconv

Libiconv converts from one character encoding to another through Unicode conversion.

Author: Bruno Haible

Homepage: http://www.gnu.org/software/libiconv/

The libiconv library is distributed and licensed under GNU Lesser General Public License version 3.

Kerio Operator includes a customized version of this library. Complete source codes of the customized version of libiconv library are available at:

http://download.kerio.com/archive/

libmbfl

libmbfl is a streamable multibyte character code filter and converter library. The libmbfl library is distributed under LGPL license version 2.

Copyright © 1998-2002 HappySize, Inc. All rights reserved.

The library is available for download at:

http://download.kerio.com/archive/

libopus

Opus is a high-quality audio codec developed in cooperation among Xiph.org, Broadcom, and Microsoft (Skype). The codec is standardized in RFC 6716. The reference implementation of the codec is licensed under a 3-clause BSD-style license. The copyright and patent licenses for the Opus algorithm are automatically granted to everyone and do not require application or approval. The patent licenses are included below together with the BSD-style license.

Copyright © 2011-2014 Opus contributors

libxml2

XML parser and toolkit.

Copyright © 1998-2003 Daniel Veillard. All Rights Reserved.

Copyright © 2000 Bjorn Reese and Daniel Veillard.

Copyright © 2000 Gary Pennington and Daniel Veillard

Copyright © 1998 Bjorn Reese and Daniel Stenberg.

nginx-nchan

Fast, horizontally scalable, multiprocess pub/sub queuing server and proxy for HTTP, long-polling, Websockets and EventSource (SSE), powered by Nginx. https://nchan.slact.net/

Written by Leo Ponomarev (slact) 2009-2015.

nginx-upload-module

A module for nginx web server for handling file uploads using multipart/form-data encoding (RFC 1867). http://www.grid.net.ru/nginx/upload.en.html

Copyright © 2006, 2008, Valery Kholodkov

OpenLDAP

Freely distributable LDAP (Lightweight Directory Access Protocol) implementation.

Copyright © 1998-2007 The OpenLDAP Foundation

Copyright © 1999, Juan C. Gomez, All rights reserved

Copyright © 2001 Computing Research Labs, New Mexico State University

OpenSSL

An implementation of Secure Sockets Layer (SSL v2/v3) and Transport Layer Security (TLS v1) protocol.

This product includes software developed by the OpenSSL Project for use in the OpenSSL Toolkit (http://www.openssl.org/).

This product includes cryptographic software written by Eric Young.

This product includes cryptographic software written by Tim Hudson.

PHP

PHP is a widely-used scripting language that is especially suited for Web development and can be embedded into HTML.

This product includes PHP software, freely available from http://www.php.net/software/

php-ev

ev provides interface to libev library - high performance full-featured event loop written in C.

PHP-JWT

A simple library to encode and decode JSON Web Tokens (JWT) in PHP, conforming to RFC 7519.

pjproject

Asterisk fork of PJSIP

ScoopyNG

This product includes software developed by Tobias Klein.

SIP.js

A simple, intuitive, and powerful JavaScript signaling library http://sipjs.com

tftpd

TFTP daemon. TFTP is a simple protocol used for file transmission.

uwsgi

uWSGI application server container http://projects.unbit.it/uwsgi

WAVPlayerProject

WAV player.

Denis Kolyako May 28, 2007, see http://etcs.ru/copyright/

zlib

General-purpose library for data compressing and decompressing.